NetGate FXO Gateway SIP User Manual (2/4/6 FXO)

FXO-02/FXO-04/FXO-06

Steps in configuration

STEP 1 Start Up	To check out the peripheral equipments and understand the feature of this gateway. Please read this step very carefully before starting the configuring.
STEP 2 How to Setup and Connect basically	Connecting the gateway and computer to start configuring by WEB GUI. Setting the ip address for this gateway to make sure that it could connect with the internet. Setting the configurations of dialing, including the Peer-To-Peer, GK mode and how to set these tables to make calls by this gateway easily. The other configurations of make call will be discussed in this step.
STEP 3 Advanced	Advanced configurations and special functions of this gateway. Using the WEB GUI to show how to set this table and explain the meaning of these tables.
STEP 4 Command List	To explain the meaning of the command in the command line interface and example the usage of the command. To get more usages or configuration in this step and study about the command line configuration.

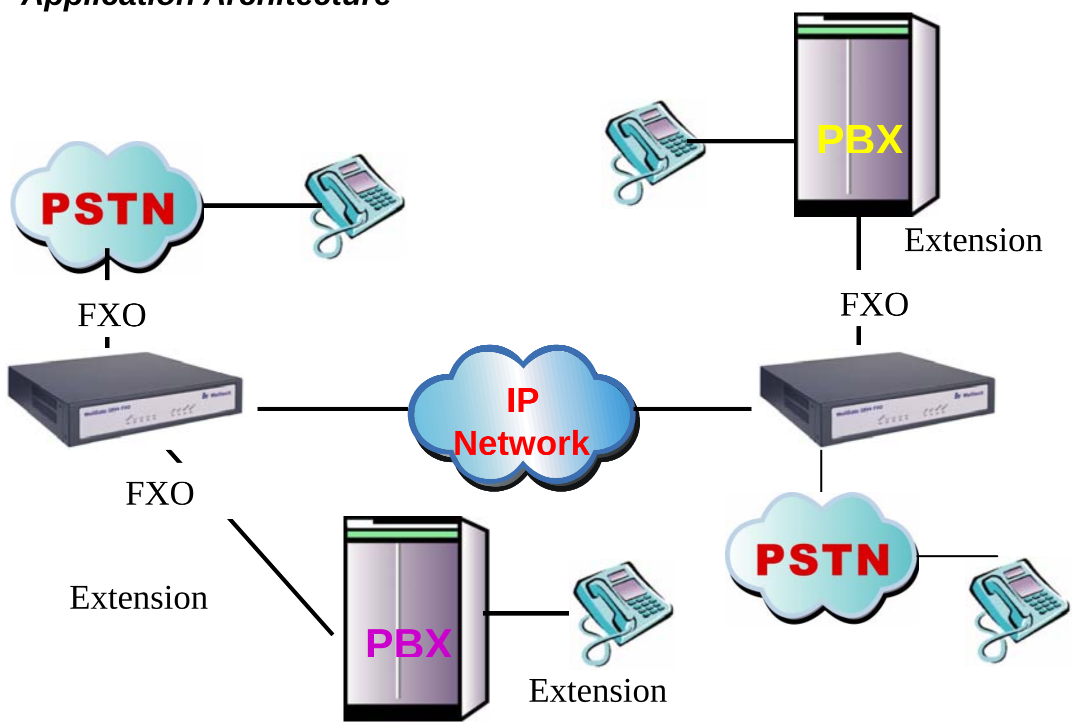
1. Start Up

1.1 Introduction

The FXO gateway provides voice/fax service over IP network with H.323 v3 protocol. By connecting to your existing ADSL or cable modem service, which allows the use of a single, network for voice and fax services with consequent saving in network infrastructure and greatly reduced telephone charges. Ideal solution for providing low cost communications between headquarters and branch offices in the world, as well as for SOHO and office telephony applications.

FXO gateway provides analog lines to connect local PSTN/PTT interface (FXO), and converts voice/fax signal onto IP network. The management feature is via RS-232C COM port and TELNET.

Application Architecture



- ***FXO ports can connect with PSTN Line or Extension Line of PBX***

1.2 Features and specification

Features

- ITU-T H.323 v3 compliance
- Automatically Gatekeeper Discovery
- Peer-to-Peer mode (non-Gatekeeper)
- Support auto-attendant (2nddial Tone / Voice greeting)
- Dimensions : 221mm(W)*42mm(H)*217mm(L)
- Line hunting
- 2(2FXO gateway)/4(4FXO gateway)/6(6FXO gateway) RJ-11 FXO ports
- E.164 (Telephone Number Plan)
- DTMF dialing
- DTMF detection/generation
- TFTP software upgrade
- Remote configuration/reset via Telnet
- LED indication for system status
- LAN interface : One RJ-45 connector of 10Base-T
- Microsoft Netmeeting v3.0 compatible
- Support static IP and DHCP
- QoS by ToS (Type Of Service)
- SNTP (Simple Network Time Protocol)
- Security: Password setting

Audio feature

- Codec -- G.711 a/μlaw, G.723.1 (6.3K/bps), G.729A (Optional)
- VAD (Voice Activity Detection), CNG (Comfort Noise Generate)
- G.168/165-compliant adaptive echo cancellation
- Dynamic Jitter Buffer
- Bad Frame Interpolation
- Gain Settings
- Provide Call Progress Tone: Dial tone, busy tone, call-holding tone and ring-back tone

Management Features:

Two easy ways for system configuration

- Console port: RS-232C port
- TELNET
- HTTP Brower (e.g. Internet Explorer)

Management Feature

- TELNET/Console port and Web Browser configuration

Certification

- UL, CE, FCC

FXO Features

- 2-wire loop start
- Support auto-attendant (Tone or voice greeting)
- PSTN polarity reversal detection
- Provide 2nd dial tone to PSTN
- Disconnect tone detection
- Asking ping function with the incoming calls from PSTN side
- Record and analyze the Tone from PSTN side

Environmental

- Operation temp:0°C to 40°C
- Humidity: 10% to 90% (Non-condensing)

1.3 Accessories and equipment

- The voice gateway in 2, 4 or 6 FXO ports and with one RJ-45 connector.
- The AC adapter.
- The CD of user manual.
- The connection cable in RS-232 interface.

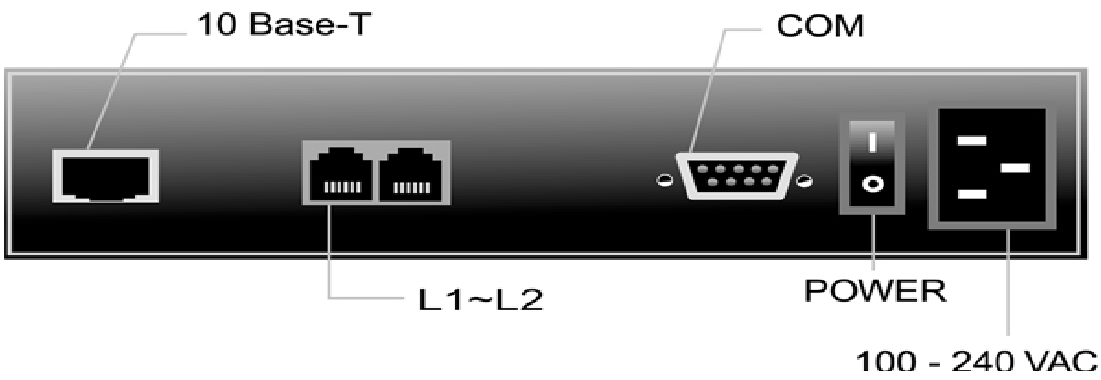
1.4 Appearance

1.4.1 2 FXO Gateway

Front panel: The LED light provides system message of 2FXO gateway.

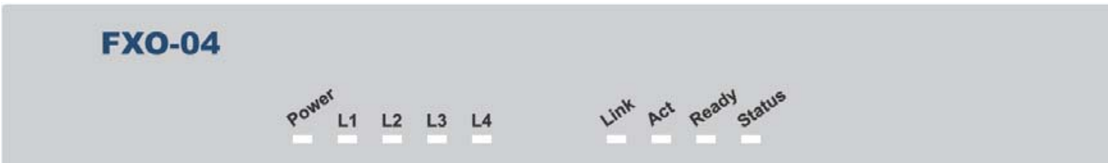
Power : Light on means 2FXO gateway is power on.
L1-L2 : Light on means the line is in use.
Link : Light on means 2FXO gateway is connected to the network correctly.
Act : LED should be light on and in flash display when data is transmitting.
Ready : 1. Light on and in slow flash means 2FXO gateway is in operation mode.
Status : 1. Light on means 2FXO gateway successfully registered to Gatekeeper when it is set as Gatekeeper Mode.
2. LED flash means 2FXO gateway is not registered to Gatekeeper when it is set as Gatekeeper Mode.
3. Or when 2FXO gateway is in downloading mode, LED should be flash as well.
4. Light off means 2FXO gateway is in Peer-to-Peer Mode.

Back panel:



1.4.2 4 FXO Gateway

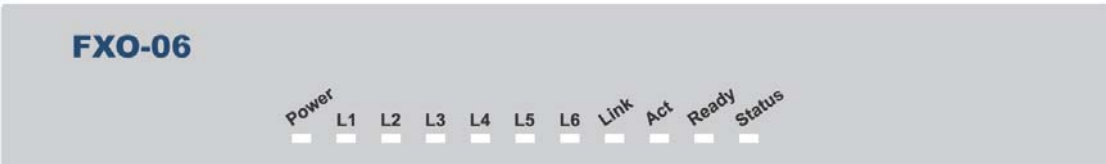
Front panel: The LED light provides system message of FXO Gateway.



Power : Light on means FXO gateway is power on.
L1-L4 : Light on means the line is in use.
Link : Light on means FXO gateway is connected to the network correctly.
Act : LED should be light on and in flash display when data is transmitting.
Ready : 1. Light on and in slow flash means FXO Gateway is in operation mode.
Status : 1. Light on means FXO Gateway successfully registered to Gatekeeper when it is set as Gatekeeper Mode.
2. LED flash means FXO Gateway is not registered to Gatekeeper when it is set as Gatekeeper Mode.
3. Or when FXO Gateway is in downloading mode, LED should be flash as well.
4. Light off means FXO Gateway is in Peer-to-Peer Mode.

1.4.3 6 FXO Gateway

Front panel: The LED light provides system message of 6FXO gateway.

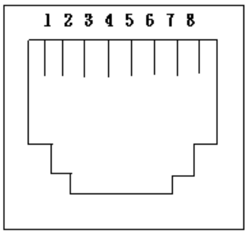


Power : Light on means 6FXO gateway is power on.
L1-L6 : Light on means the line is in use.
Link : Light on means 6FXO gateway is connected to the network correctly.
Act : LED should be light on and in flash display when data is transmitting.
Ready : 1. Light on and in slow flash means 6FXO gateway is in operation mode.
Status : 1. Light on means 6FXO gateway successfully registered to Gatekeeper when it is set as Gatekeeper Mode.
2. LED flash means 6FXO gateway is not registered to Gatekeeper when it is set as Gatekeeper Mode.
3. Or when 6FXO gateway is in downloading mode, LED should be flash as well.
4. Light off means 6FXO gateway is in Peer-to-Peer Mode.

1. **Ethernet Port**

   LAN/WAN: 10/100 Base-T; RJ-45 socket, complied with ETHERNET 10/100base-T.

   The pin-out is as following:



PIN 1, 2: Transmit

PIN 3, 6: Receive

2. **COM:**

   RS232 console port (DB-9pin **male** connector)

   Note: use straightforward cable to connect to your computer.

PINOUTS

Pin	Name	Dir	Description
2	RXD		Receive Data
3	TXD		Transmit Data
5	GND		System Ground

3. **LINE:**

   RJ-11 connector, FXO interface is for connecting the extension line of PABX or PSTN Line.

4. **12V DC:**

   Input AC 100V~120V;output DC12V.

2. How to Setup and connect basically

2.1 System Requirement

1. One PC (a) Pentium 100 or above, 64 RAM, Windows 98 or above.
   (b) Ethernet card or COM port
2. One standard straightforward RS-232 cable (female connector to Gateway side).
3. PBX extension Lines or PSTN Lines.
4. Software tools (a) Hyper Terminal, TELNET, Web Browser.
   (b) Gatekeeper (optional).

2.2 IP Environment Setting

User must prepare a valid IP address, complied with IP Network, for Gateway's proper operation.

For testing the validation of chosen IP address, using the same IP configuration in other PC or Notebook, and then try to connect to Public Internet (go to well-known website, receive Internet mail, or ping a specific public IP address). If it works, use the same IP address and network configuration for Gateway.
Please follow up the step for the configuration of your computer or notebook.

2.2.1 For Windows 2000/NT

Please make sure that the network interface of your computer is working fine and the cross over line (RJ-45) is connecting with the computer correctly or you could use a hub to connect with your computer and this gateway. Turn on your computer and configure the network parameter as follow:

1 Go to the **start** menu and enter the **setting** area. Click **control panel**.

2 Enter the network configuration.

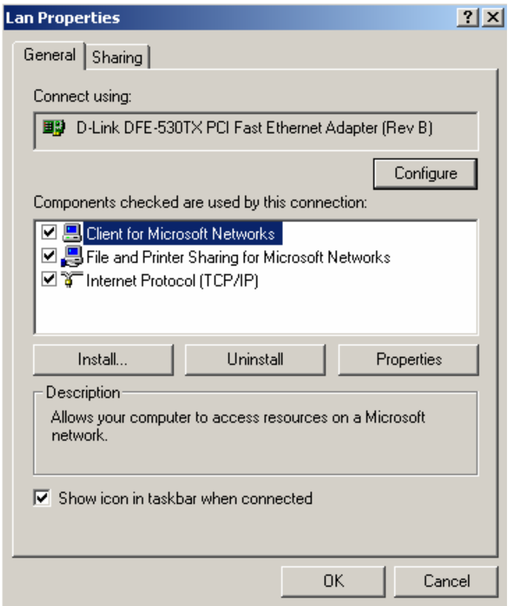


Figure 2.1: Network Configuration

3 Select the **Property** of the LAN card.

4 Setup the ip address, subnet mask and default gateway as below:

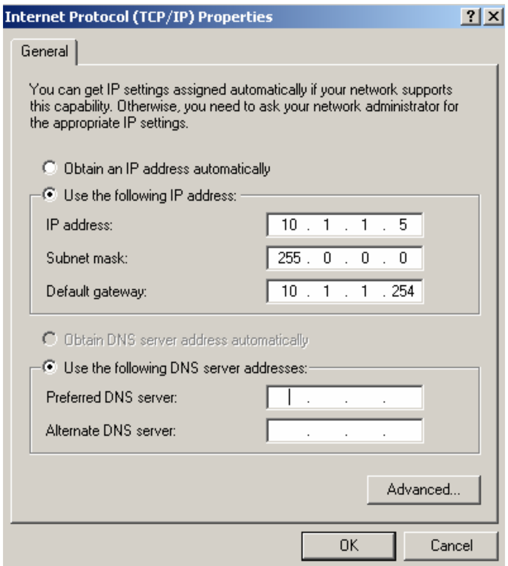


Figure 2.2: Configure the network

5 Click OK after you finished the network setup.

The default ip address, netmask and default gateway address of the gateway is 10.1.1.3, 255.0.0.0, 10.1.1.254.

2.3 Network configurations in your gateway

1 Key in the ip address of the gateway (http://10.1.1.3) with the browser. (see figure 2.3)

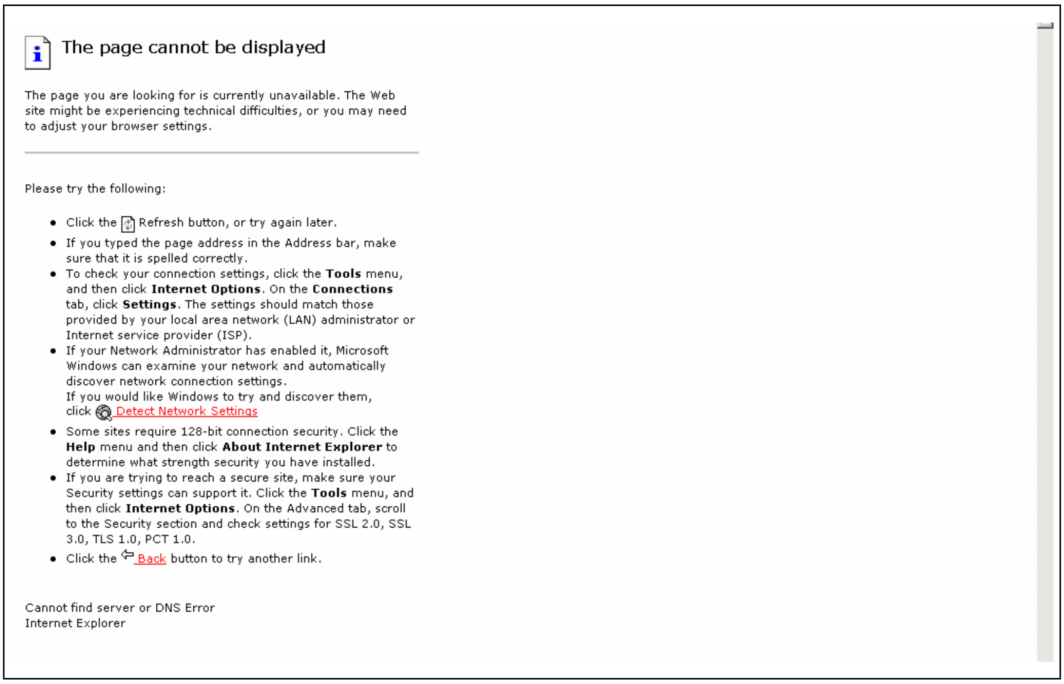


Figure 2.3: WEB Browser

2 After key in the ip address, you have to enter the user name and password to enter the WEB configuration. (Username: root ; No password) (see figure 2.4)

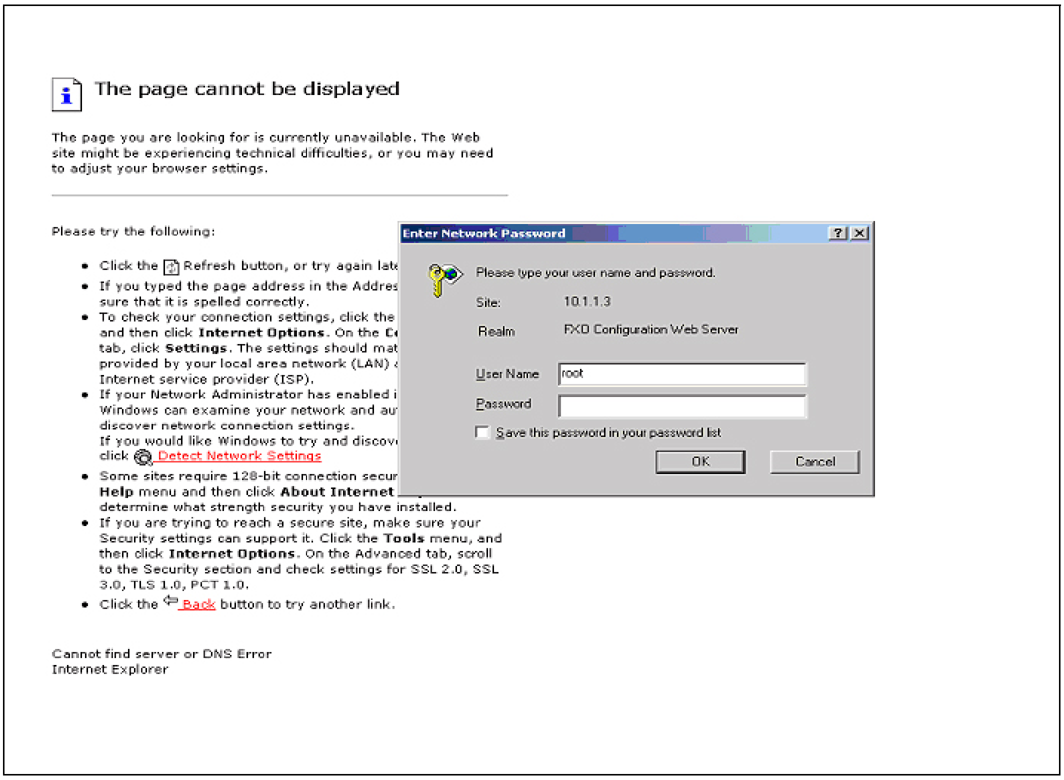


Figure 2.4: Login the username and password

3 You will enter the main page of the configuration after key in the login name and password correctly: (see figure 2.5)

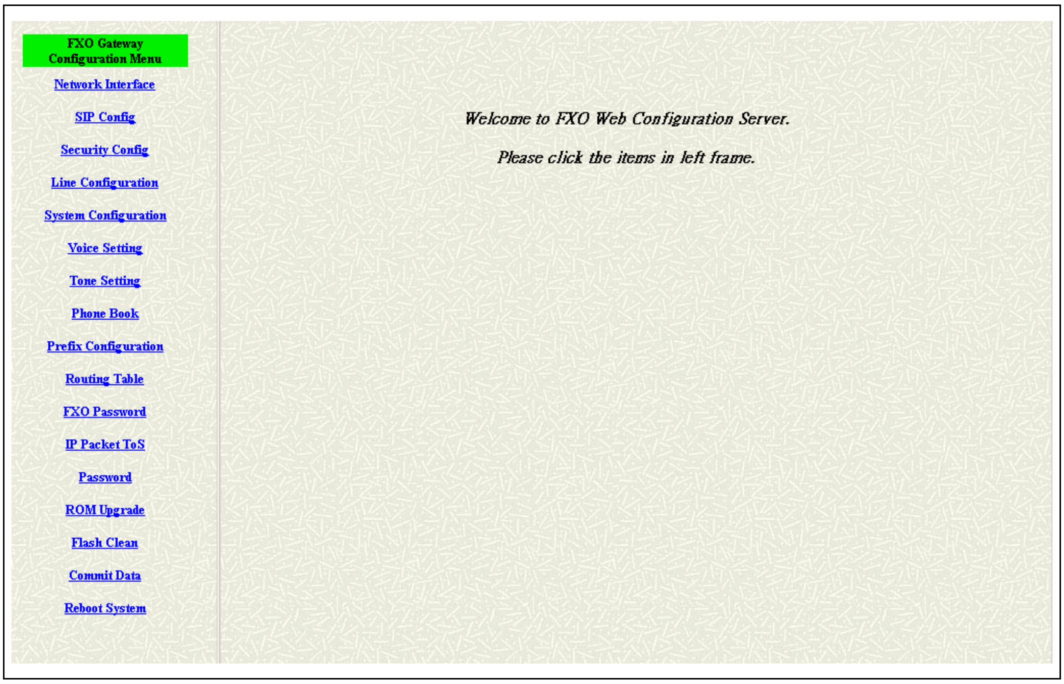


Figure 2.5: The main WEB configuration

4 Press the **Network Interface** to configure the networking of your gateway. (see figure 2.6)

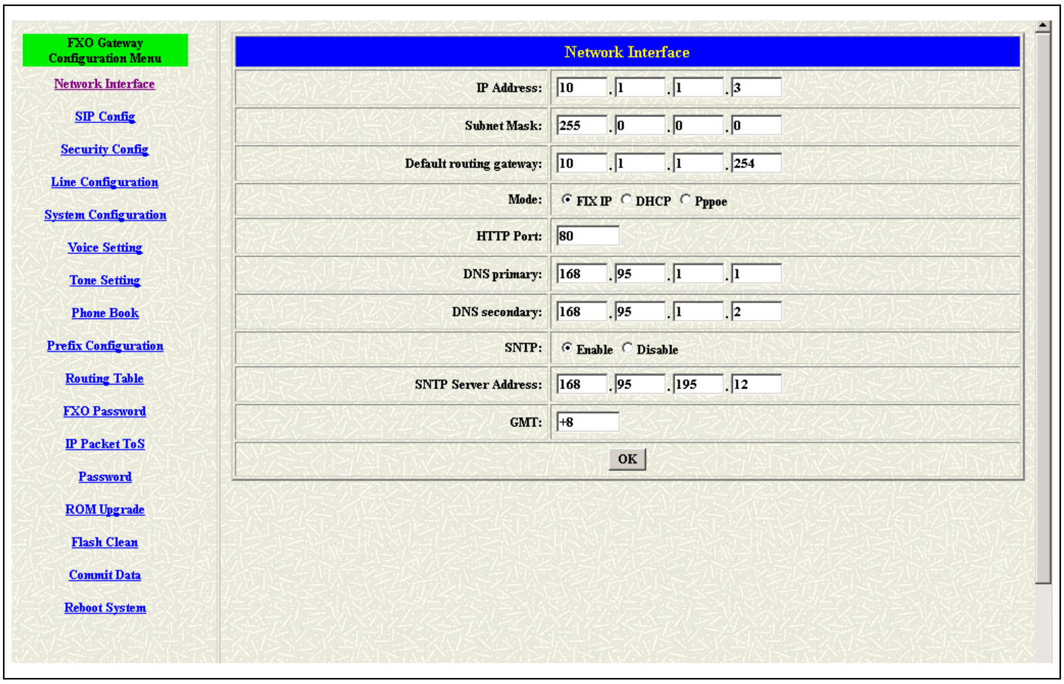


Figure 2.6: The Network Interface

2.3.1 Static ip address

1 Please get the correct ip address, netmask and default gateway address from your ISP first. Press the OK button if you finished. (see figure 2.7)

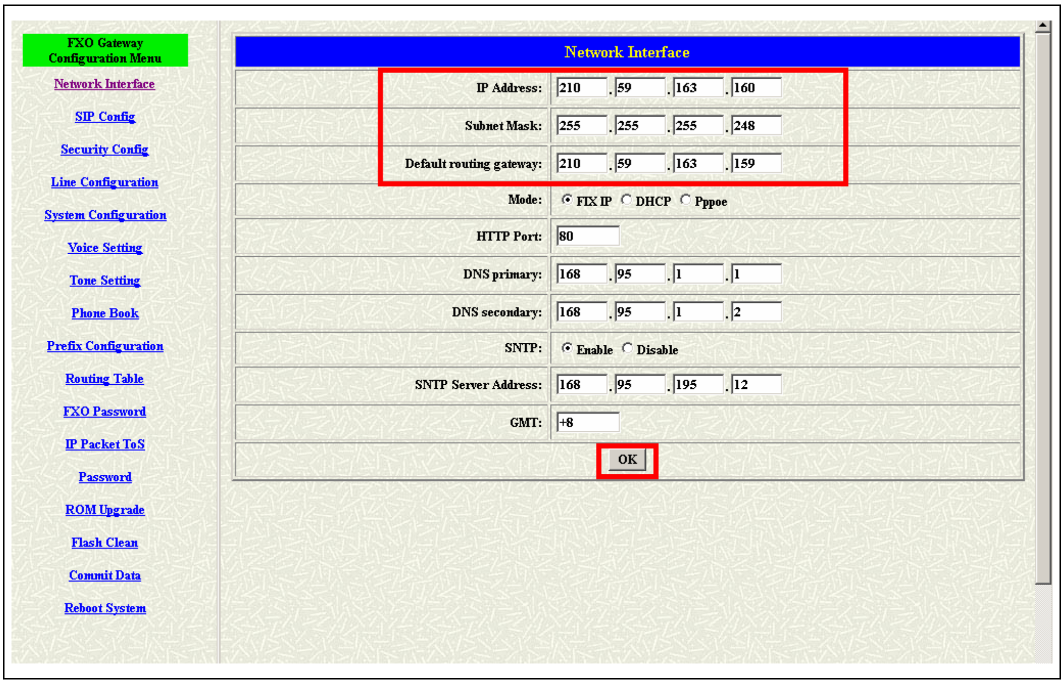


Figure 2.7: Configure the static ip address

2 Press the commit if you finish the configuration. (see figure 2.8)

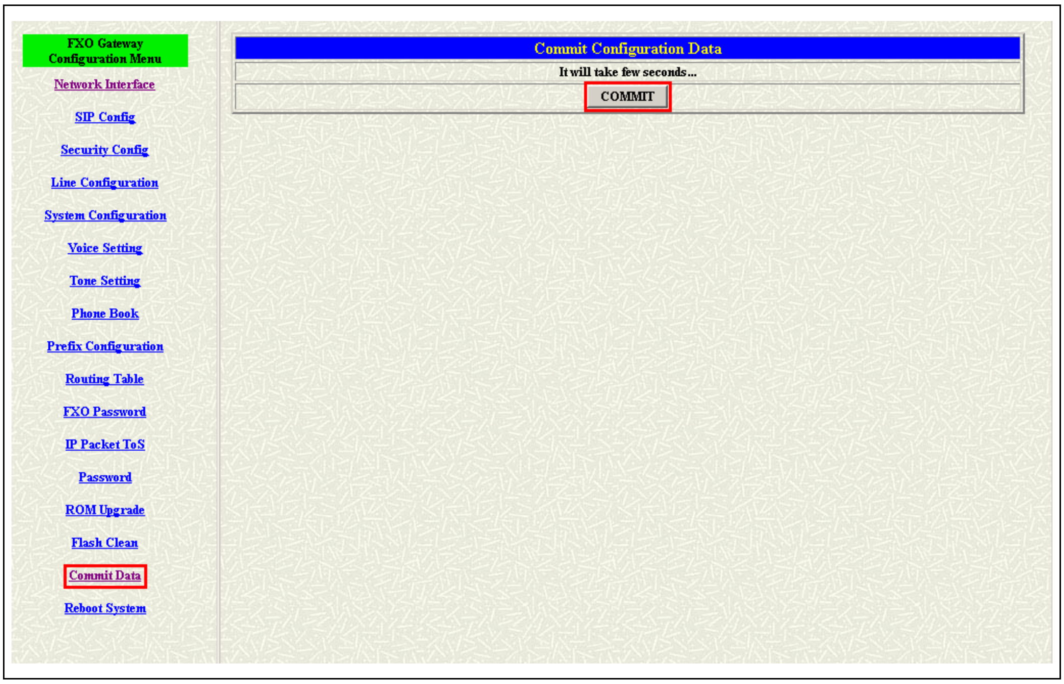


Figure 2.8: Commit the data

3 Press the reboot if you want the configuration executed. (see figure 2.9)

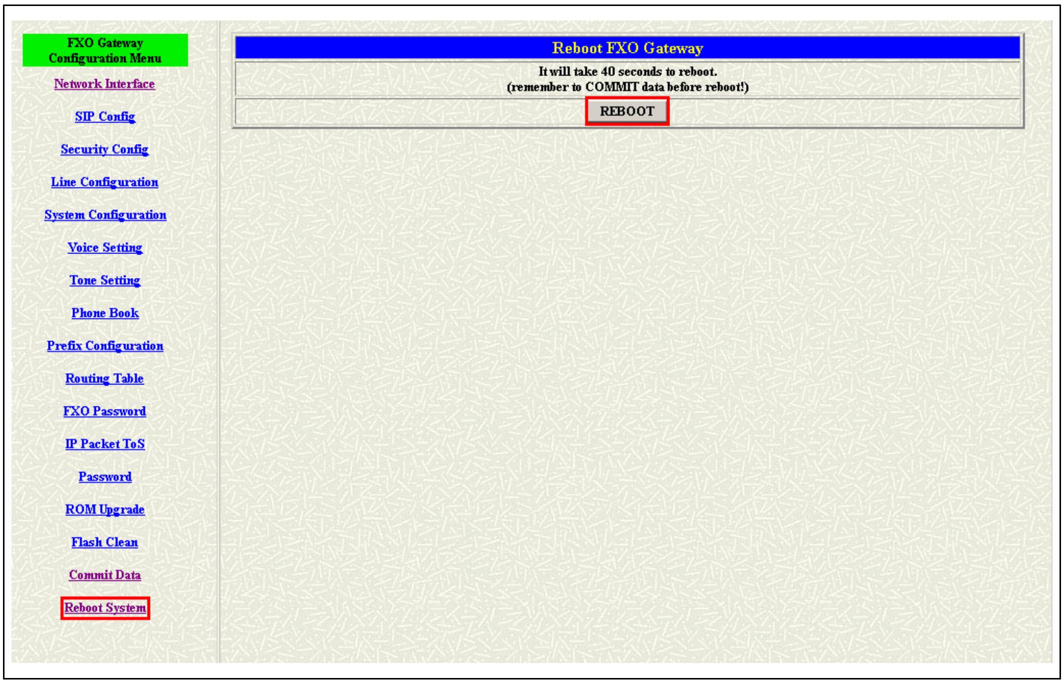


Figure 2.9: Reboot the system

2.3.2 DHCP mode

1 Enable the DHCP if you are using the cable modem or DHCP server. (see figure 2.10)

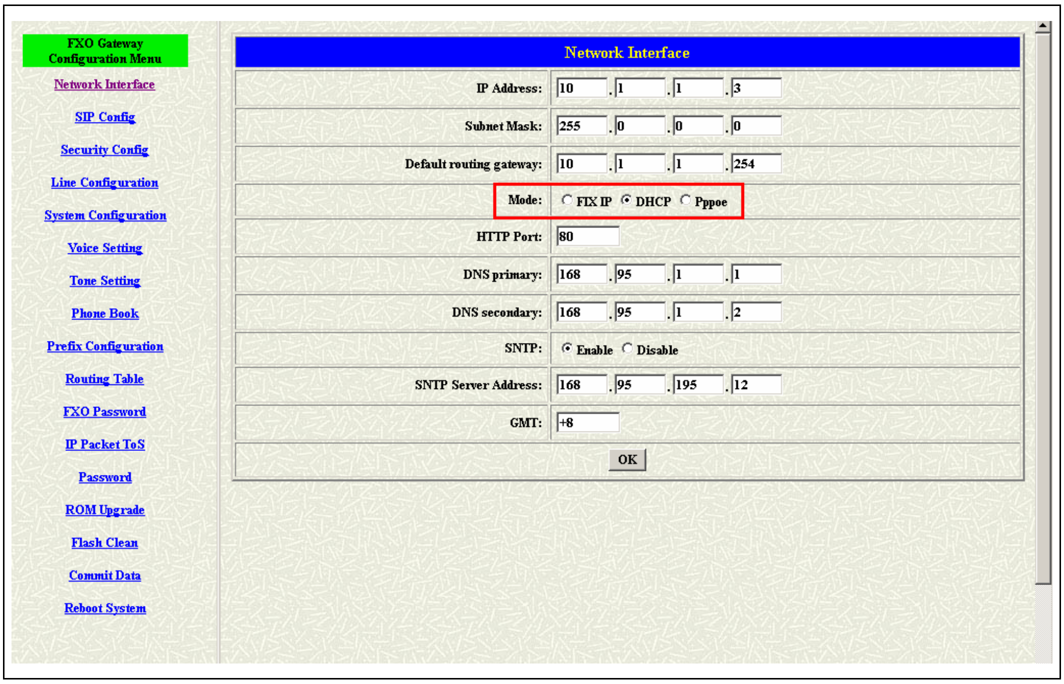


Figure 2.10: Enable the DHCP function

2 Please commit the data and reboot the machine after you enable the DHCP function.

2.3.3 PPPoE mode

1 Switch to the PPPoE mode and press the “OK” button. Press the **Network Interface** button after the “OK” button. (see figure 2.11)

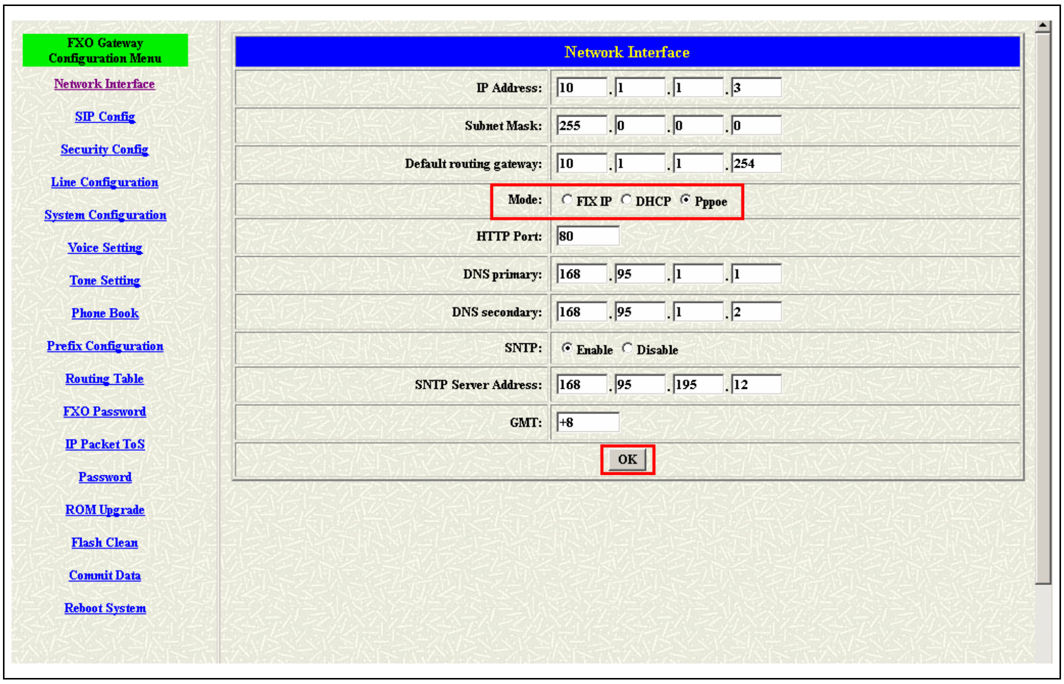


Figure 2.11: Switch to the PPPoE mode

2 Enter the Login account and password. Press the “OK” button if the configuration is finished. (see figure 2.12)

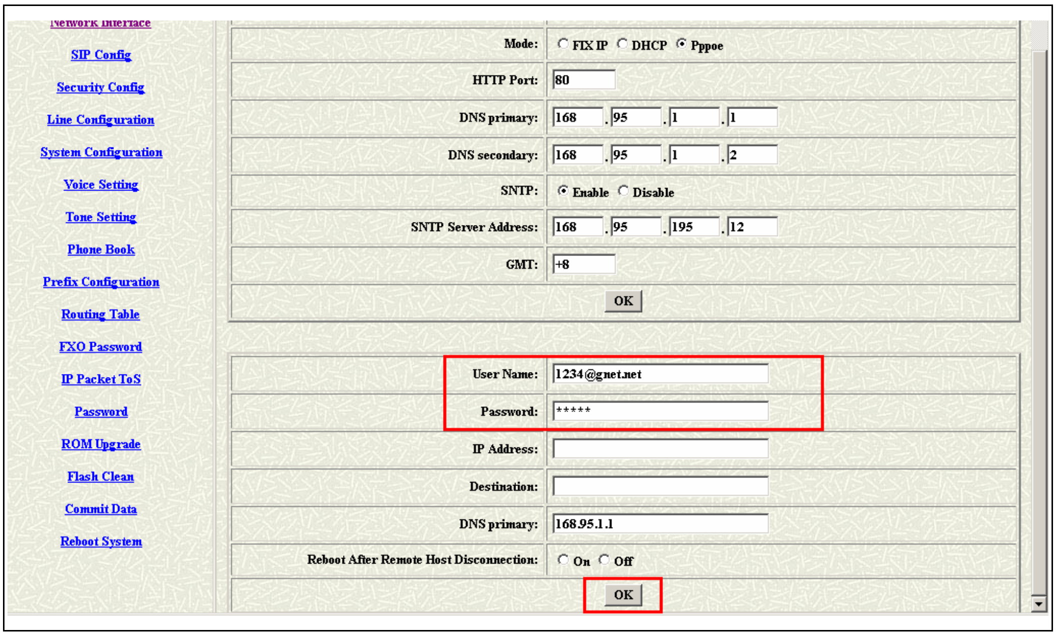


Figure 2.12: Enter the Account and password

2 Please commit the data and reboot the machine after you finished the configuration about the PPPoE function.

2.4 Making a VoIP Call

There are two modes that you could configure the gateway for making VoIP calls. One is the Peer-to-Peer mode, another is Proxy mode. The configurations and functions are different. Please make sure about the mode you want and follow up the step to configure your gateway.

2.4.1 Configure the gateway into the Peer-to-Peer mode

1 Enter the SIP Configuration table and change the mode to Peer-to-Peer. Define the port numbers whatever you like. Press the “OK” button if the configuration is all finished. (see figure 2.13)

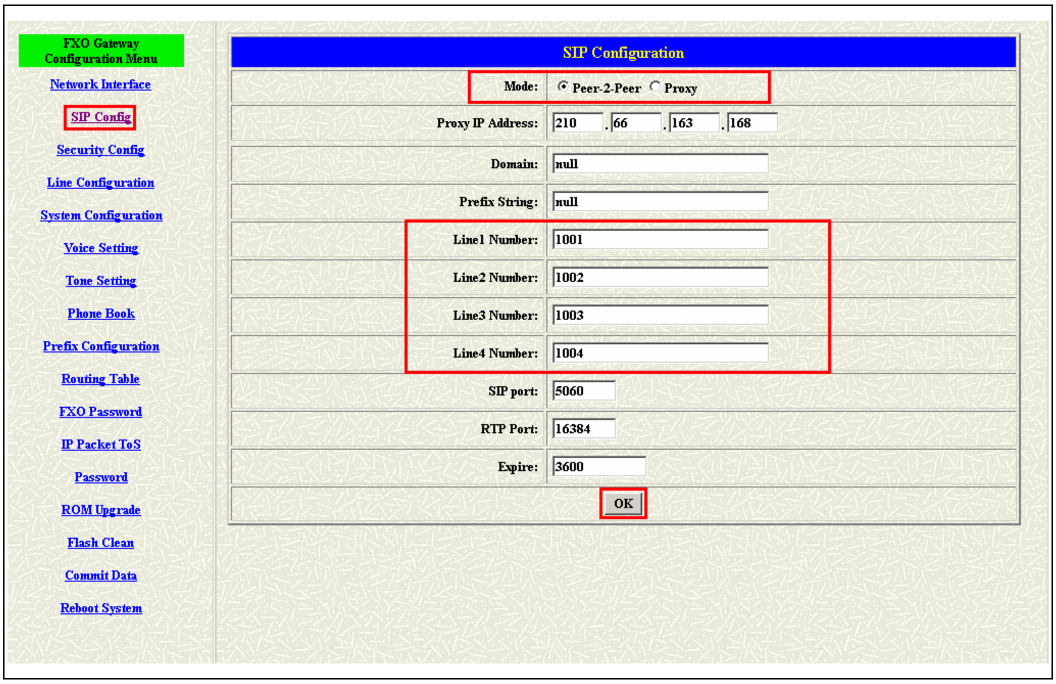


Figure 2.13: Configure the Peer-to-Peer mode

2 Enter the Phone Book configuration table and configure the name, ip address and phone number of the destination. (see figure 2.14)

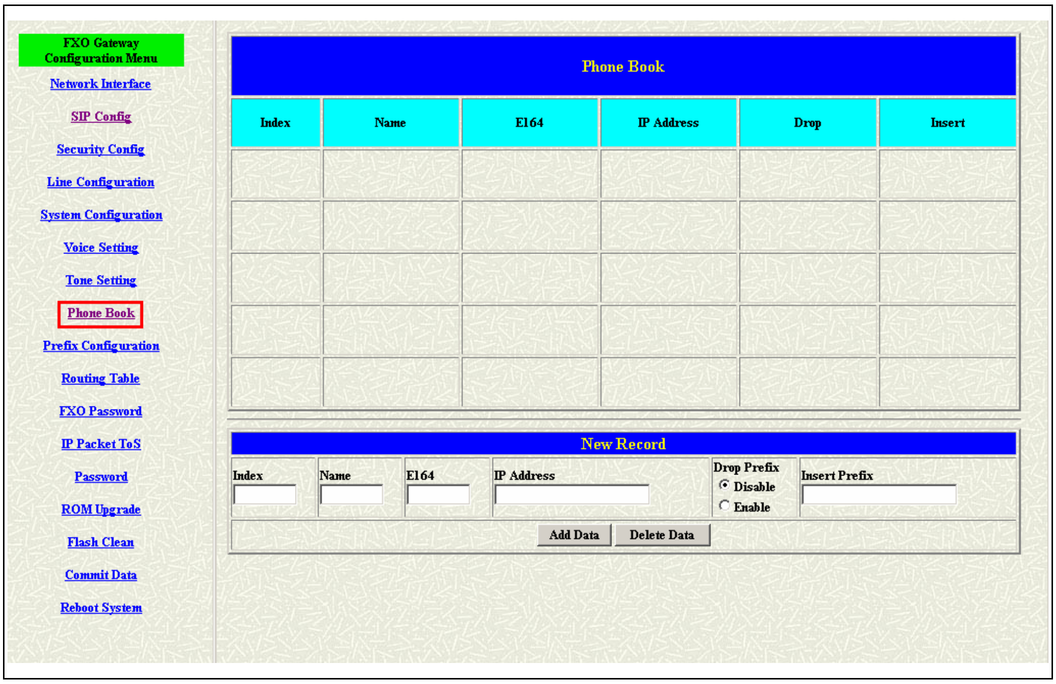


Figure 2.14: Phone Book

【Example】

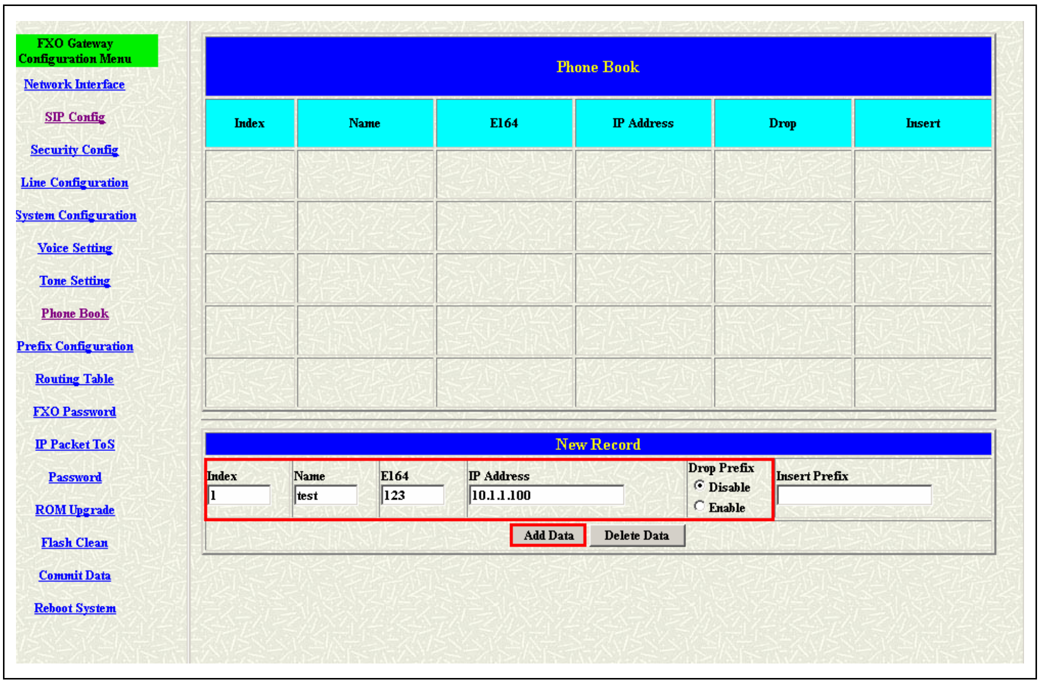


Figure 2.15: The example of Phone Book configuration

The name of the destination: **test**
The E164 number (phone number) of the destination: **123**
The ip address of the destination: **10.1.1.100**
(The port will be 1720 if you don’ t define it)
Drop prefix: **Enable – The e164 number you define will be deleted**
Disable – The e164 number you define will be kept
Insert prefix: **To add a number you define in this table**

Press the “Add Data” button when you finished, and the new table will display on the first index if you press the Phone Book configuration button.

4 Please Commit it and Reboot the system if the configuration is finished. (see figure 2.16)

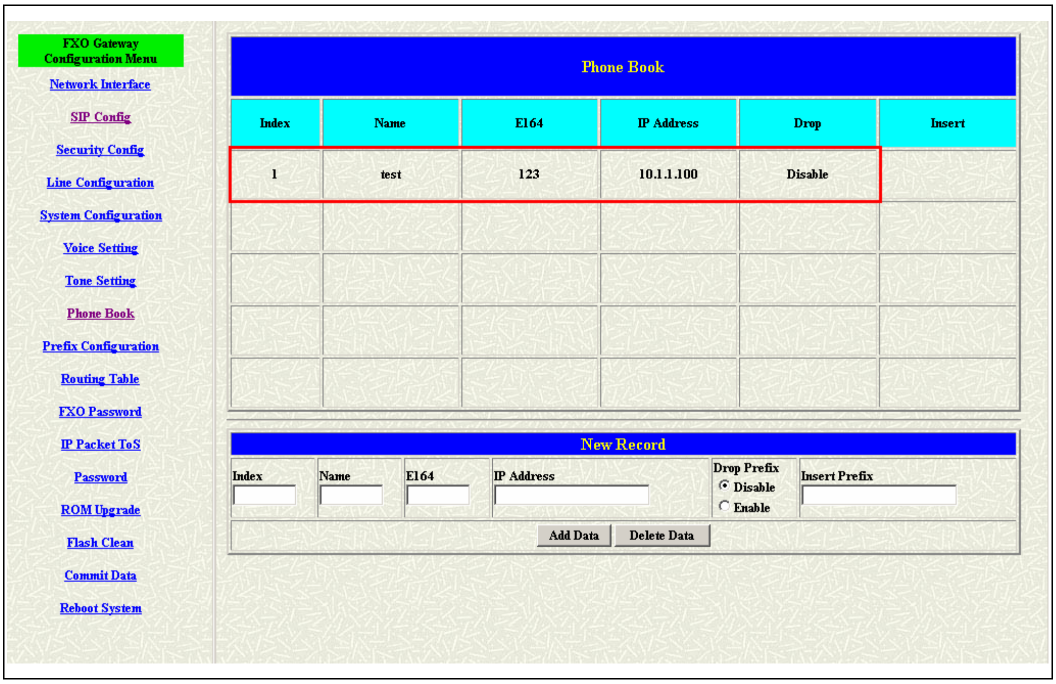


Figure 2.16: To show the Phone Book record

Phone Book is only for the Peer-to-Peer mode.
Fifty index support.

【The application in the drop and insert function】

Input (E164)	Drop	Insert	Output
100	Disable	X	100
200	Disable	0	0200
300	Enable	X	X
400	Enable	500	500

※ X – Do not enter any numbers

2.4.2 Configure the gateway into the Proxy mode

1 Enter the SIP Config table and change the mode from Peer-to-Peer to Proxy.

To change the Proxy information from your service provider (Ex: The Proxy IP, Domain and Line numbers). (see figure 2.17)

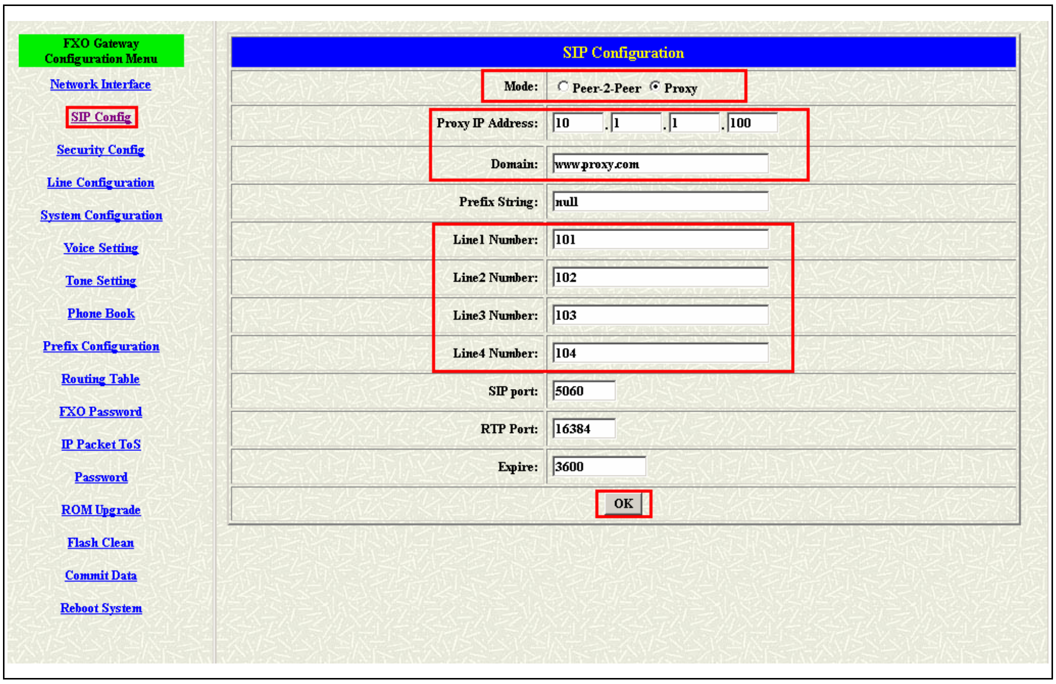


Figure 2.17: Configure the Proxy info

2 Press the OK button that is on the bottom of this page to save the configuration.

3 Switch to the Security Config page and put the user account and password in the correct table. Please get this info from your ITSP. Press the OK button if the configuration is finished. (see figure 2.20)

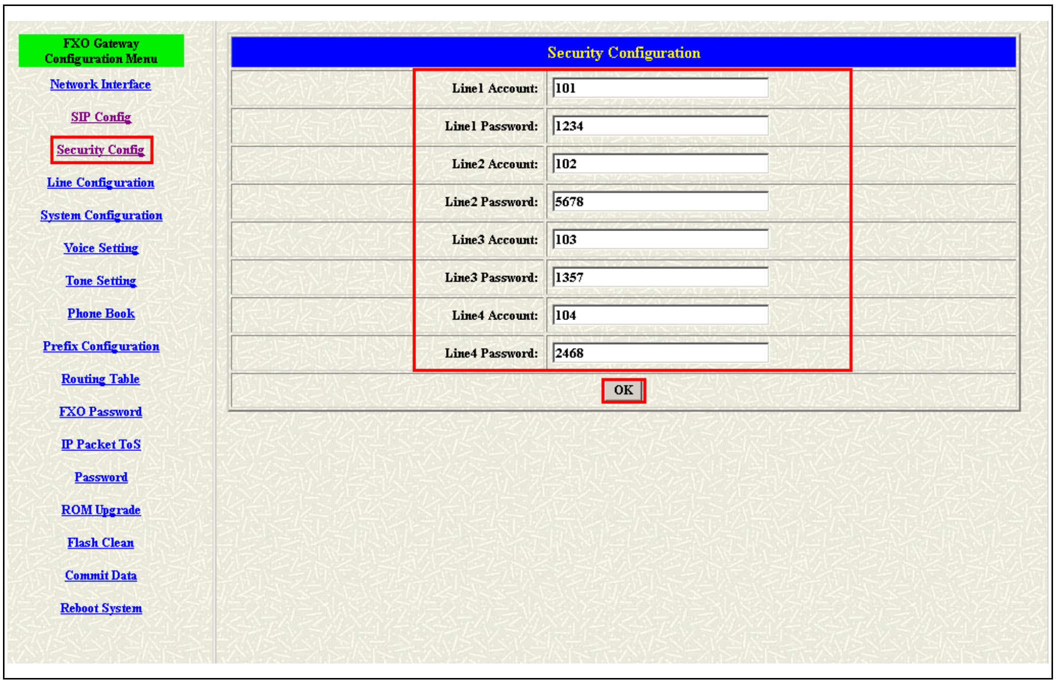


Figure 2.20: Configure the Security info

4 Press the Commit Data and Reboot System buttons when you finished the configuration.

3. Advanced

There are too many advanced commands for the advanced users. The following chapters are based on the application layer. Please get the info what you need. If you need the command, please watching the chapter of Command Line Interface.

3.1 Network Configuration

The Network configuration will help users to configure the info about the network. Please get more detail info from the following. (see figure 3.1)

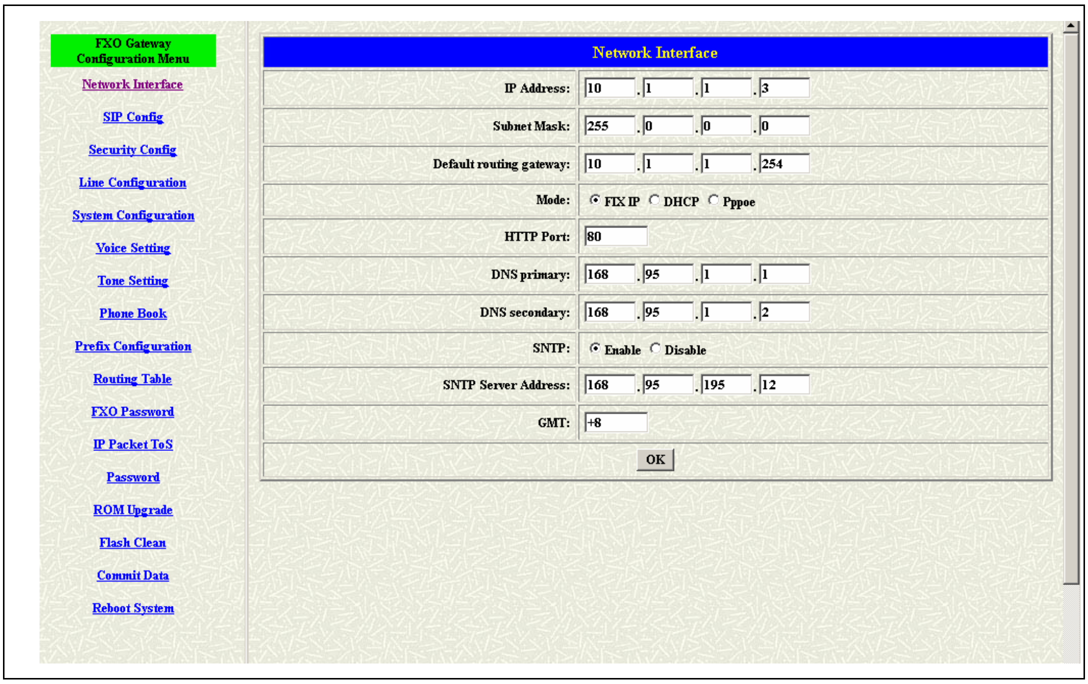


Figure 3.1: Network Configuration

- IP Address – Define the ip address for your networking if it is the fixed ip. Please get this info from your ISP.
- Subnet Mask – Define the mask address for your networking. Please get this info from your ISP.
- Default Gateway – Define the default gateway for your networking. Please get this info from your ISP.
- Mode – Users could define the networking type for this gateway. It could support the Static, DHCP and PPPoE function.
- HTTP Port – This port is for the WEB configuration. The default port for the WEB is users could change it by this table.
- DNS primary – Users could define the primary DNS server address.
- DNS secondary – Users could define the primary DNS server address.
- SNTP – Enable the SNTP server registering function if user wants to get the correct time from the Command Line Interface.
- SNTP Server Address – Enter the correct ip address of the SNTP server or get the incorrect time from the Command Line Interface.
- GMT – Configuring the time area for the time display in the Command Line Interface.

The following is for the PPPoE configuration. (see figure 3.2)

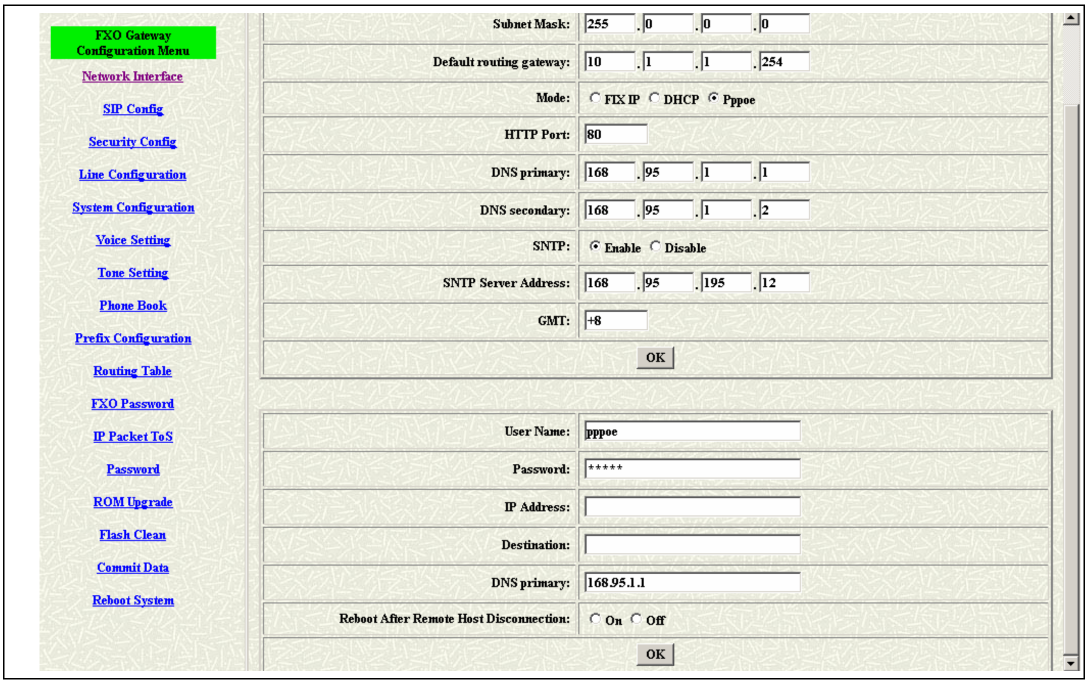


Figure 3.2: PPPoE Configuration

- PPPoE User Name – Put the PPPoE connection account in this table. Please get this info from your ISP.
- PPPoE Password – Put the PPPoE connection password in this table. Please get this info from your ISP.
- PPPoE IP Address – After the connection success, this table will show you the IP address which the gateway got from the ISP.
- PPPoE Destination – After the connection success, this table will show you the default gateway address, which the gateway got from the ISP.
- PPPoE DNS primary – After the connection success, this table will show you the DNS ip address from the ISP.
- Reboot After Remote Host Disconnection – Enable this function will make the gateway restart automatically if the PPPoE connection is disconnected or the IP address was taken back by the ISP.

3.2 SIP Configuration

For the Proxy mode, users have to put the info about the Proxy in to this configuration table and configure the phone number. (see figure 3.3)

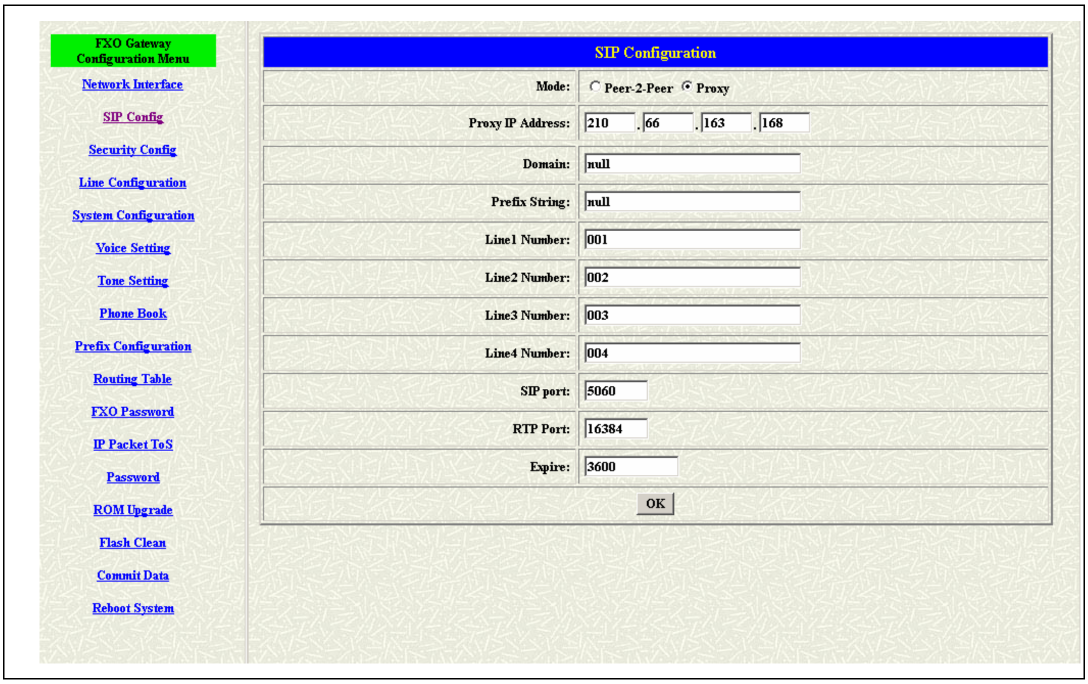


Figure 3.3: SIP Configuration

- Mode – Switch the P2P or Proxy mode.
- Proxy IP Address – Enter the IP address of the SIP Proxy.
- Domain – Enter the domain name of the SIP Proxy.
- Prefix String – For the special registration for the special proxy. This configuration could use the letters for the registration.
- Line1 Number – The phone number for the port 1.
- Line2 Number – The phone number for the port 2.
- Line3 Number – The phone number for the port 3.
- Line4 Number – The phone number for the port 4.
- SIP Port – To adjust the SIP port for this unit.
- RTP Port – The RTP port for the communication.
- Expire – The TTL time.

3.3 Security

Users could define the account and password for the port for the registration. (see figure 3.4)

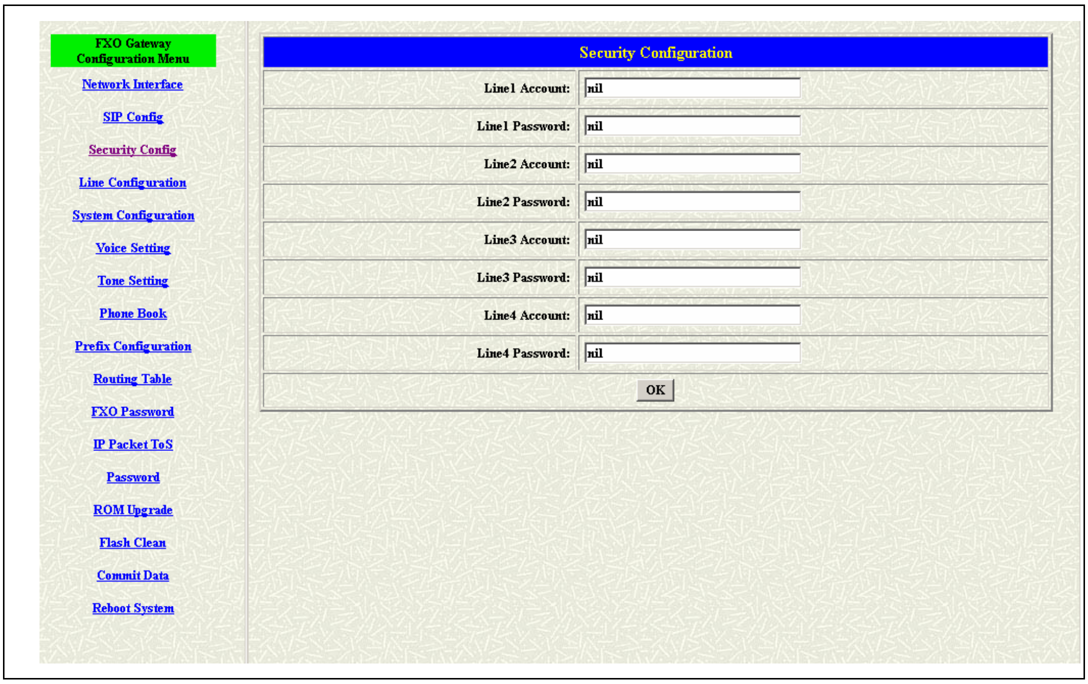


Figure 3.4: Security Configuration

- Account – The account name for this port.
- Password – The password for this account.

3.4 Line

The Line configuration will show the status of the registrations and the ports. It includes the hunt group, hotline, and no answer forward configuration. Press the Line configuration button to enter configuration table (see figure 3.5)

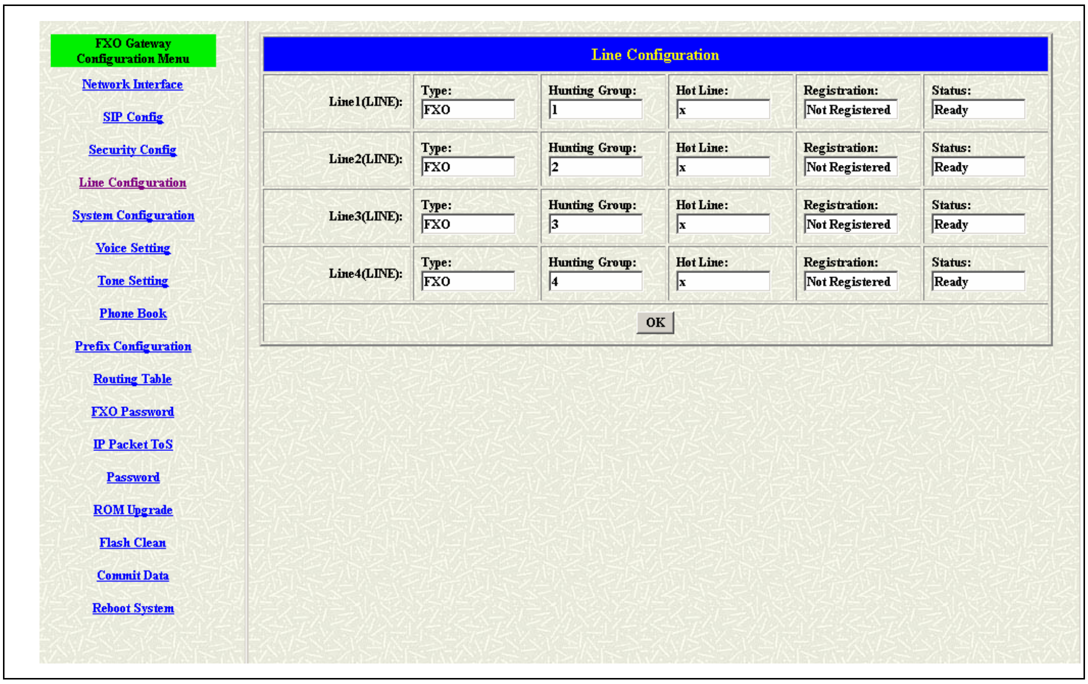


Figure 3.5: Line Configuration

- Type – Show the type of this port. There is only FXO type of this gateway, and it couldn't be changed.
- Hunting Group – Define the group number of this port. When the port is busy, the call could be transferred to another port in the same group.
- Hotline – Enable or Disable the hotline mode. The hotline mode will be enabled if you enter the hotline number. The default setting is disabled.
- Registration – Showing the gateway registered on the Proxy or not.
- Status – Showing the port is busy or ready.

3.5 System Configuration

There are some parameters in the system configurations, please get more detail as following. (see figure 3.6)

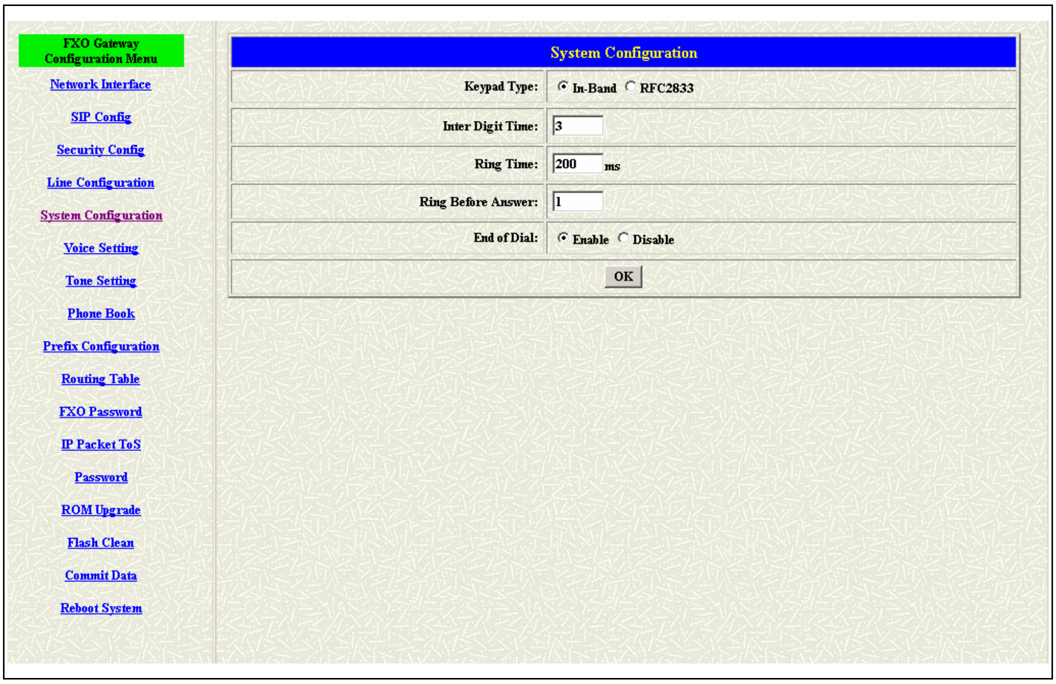


Figure 3.6: System Configuration

- Keypad type – There are two types for the Keypad. One is the In-Band type, another is the RFC2833 type. User could define the keypad type for the dialing.
- Inter Digit Time – It's the time for the time out during the dialing numbers.
- Ring Time – FXO will detect the ring tone according this time.
- End of Dial – It will transfer the digit "#" if this function was disable.

3.6 Voice Setting

User could define some parameters about the voice in this voice-setting page. (see figure 3.7)

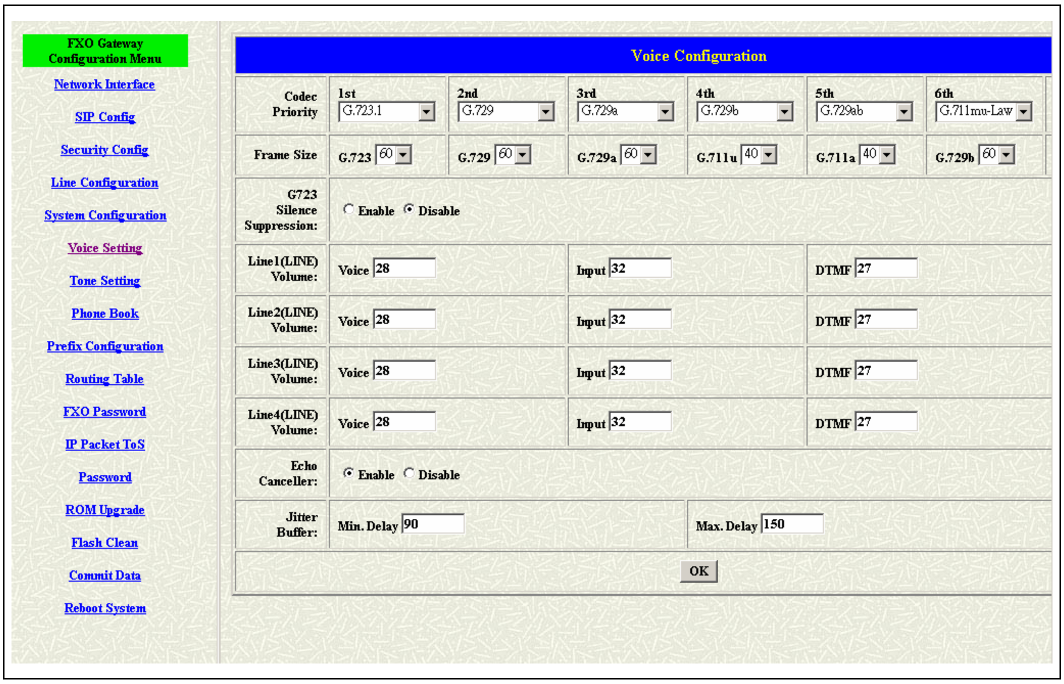


Figure 3.7: Voice Setting

- Codec Priority：It's for the codec setting. User could use the codec, which they want by the setting.
- Frame Size：It's the packet size for all codec. It will take more bandwidth if user configure the packet size in the minimum value.
- G723 Silence Suppression：For the VAD and CNG function support.
- Volume：To adjust the gain of the output, input and dtmf.
- Echo Canceller：To enable the echo cancellation function.
- Jitter Buffer：To adjust the Jitter Buffer size to avoid the packets losing.

A large jitter buffer causes increase in the delay and decreases the packet loss. A small jitter buffer decreases the delay but increases the packet loss. The size of the jitter buffer depends on the condition of the network, which varies with time. Typically the packet loss should be less than 10% for a good quality of speech.

3.7 Tone Setting

The Tone Setting is for the Tone detecting. The call will be dropped if the pattern of the tone from PSTN side is as same as the pattern in the disconnect tone table. The same result for the Ring Back Tone. User could define the pattern of the disconnect tone if the disconnect tone from PSTN side is not the standard tone. (see figure 3.8)

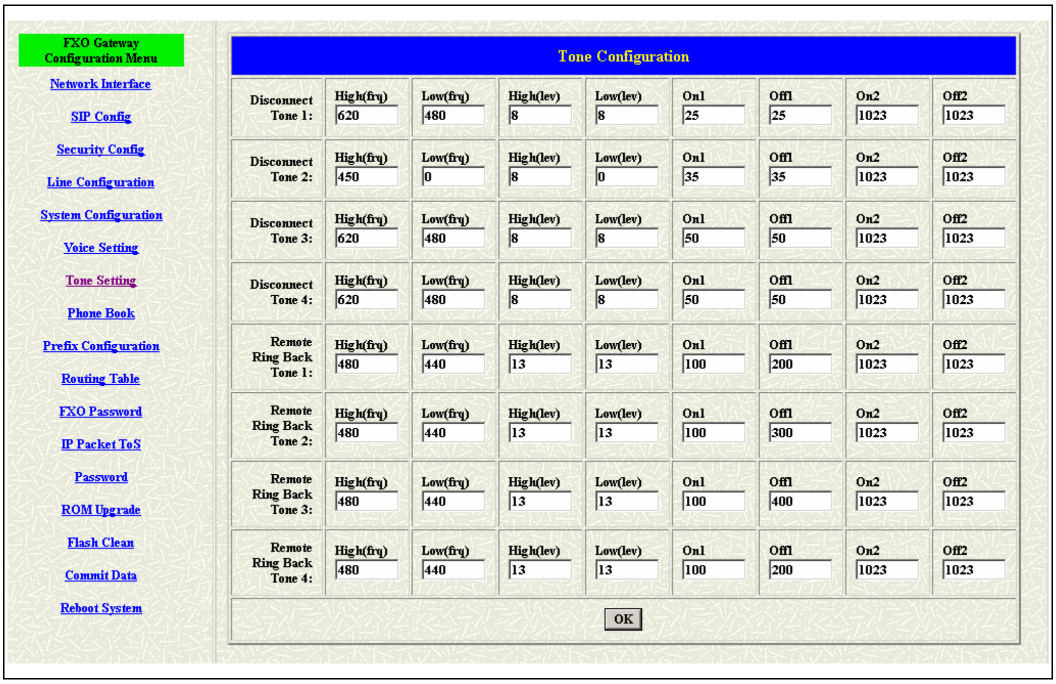


Figure 3.8: Tone Setting

- Disconnect Tone – Users could put the correct pattern of the disconnect tone in this table. The call will be dropped if the tone from PSTN side is match with these patterns. Users could have four tables for the disconnect tone.
- Remote Ring Back Tone – User could adjust this table to help this gateway to detect the Remote Ring Back Tone. There could be four tables for the configuration.

3.8 Phone Book Configuration

The Phone Book could only support the Peer-to-Peer mode. Users have to put the complete info in this table and follow up the E164 number from this table. Please get the detail info from the following. (see figure 3.9)

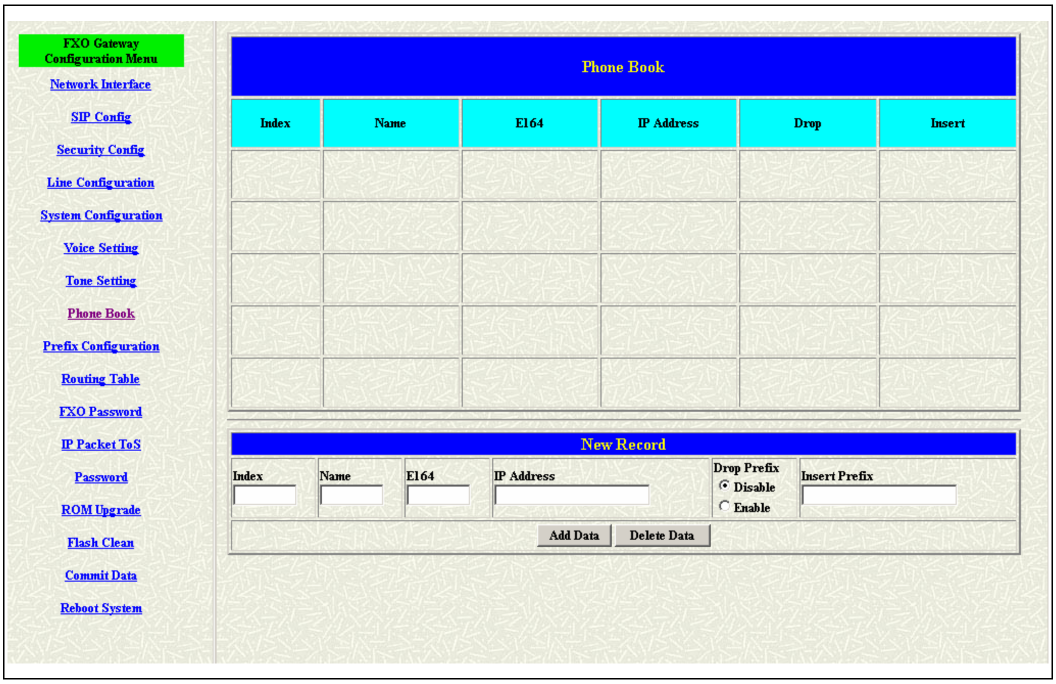


Figure 3.9: Phone Book Configuration

- Index – The number for this record.
- Name – The name for this record.
- E164 – The dialing number for this record.
- IP Address – The IP address for this destination.
- Drop – For the drop digits function.
- Insert – For the insert digits function.
- Add Data – Users have to put the whole info about this record and press this button to add this in the table.
- Delete Data – Users have to put the index number in the index table and press this button to delete this record.

3.9 Prefix

The Prefix function is using the drop and insert function (see figure 3.10).

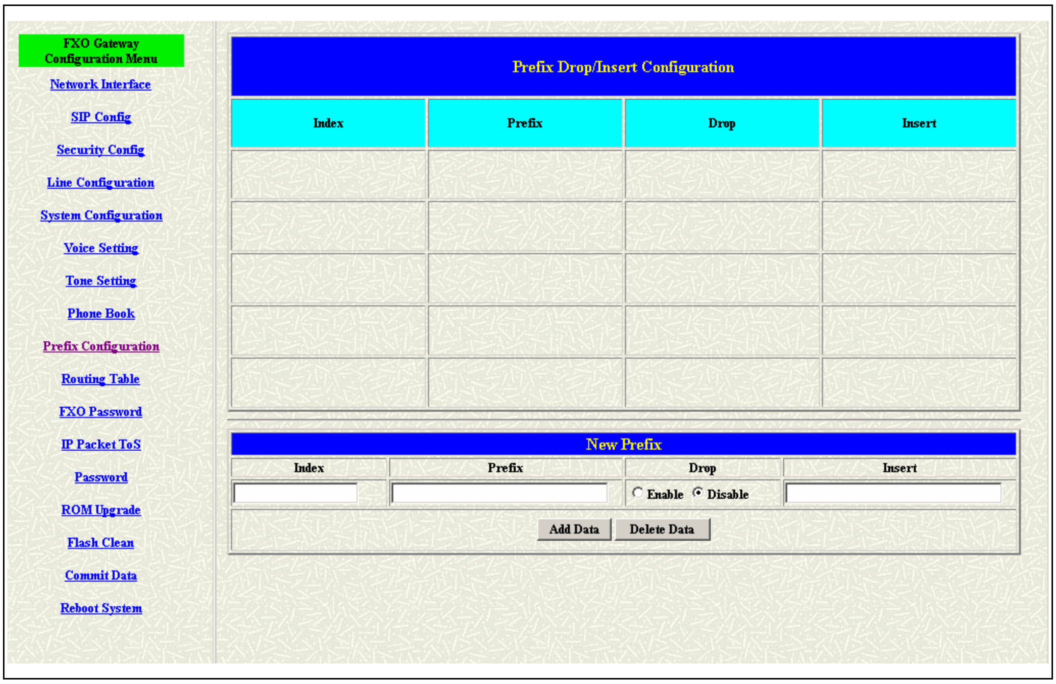


Figure 3.10: Prefix Configuration

There is a rule between Prefix and Routing command, the Prefix command have the higher priority over the Routing command. If there is an incoming call from any sides, the Routing will check this calling number after the Prefix checked (see figure 3.)

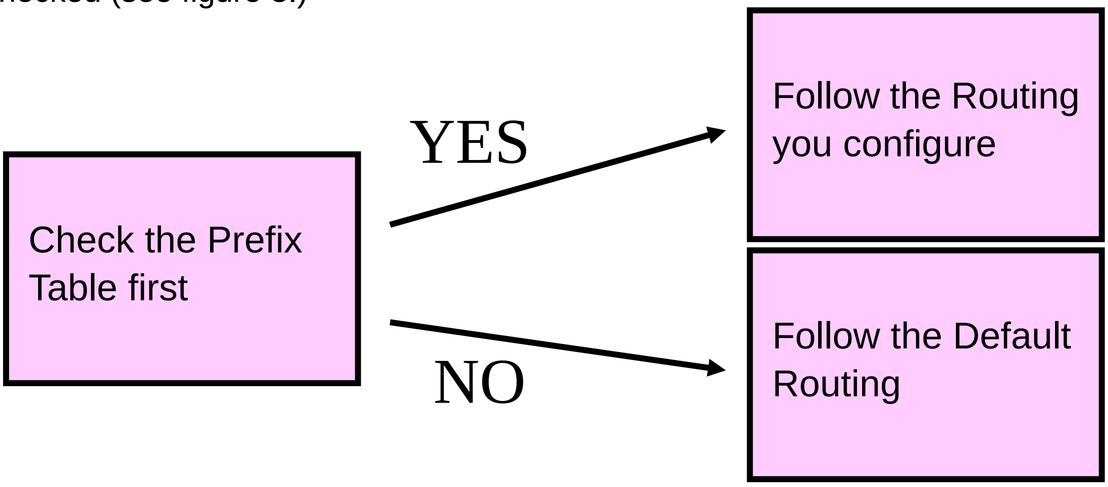


Figure 3.11: The Priority

There is an example about the configuration, please follow up these steps.

1 Press the Prefix Configuration button to enter the configuration table (see figure 3.6)

2 Enter the index number. Put the prefix numbers you will dial in the prefix table, enable (disable) the drop function and enter the numbers you want to insert. Pressing the add data button to add this record. (see figure 3.12).

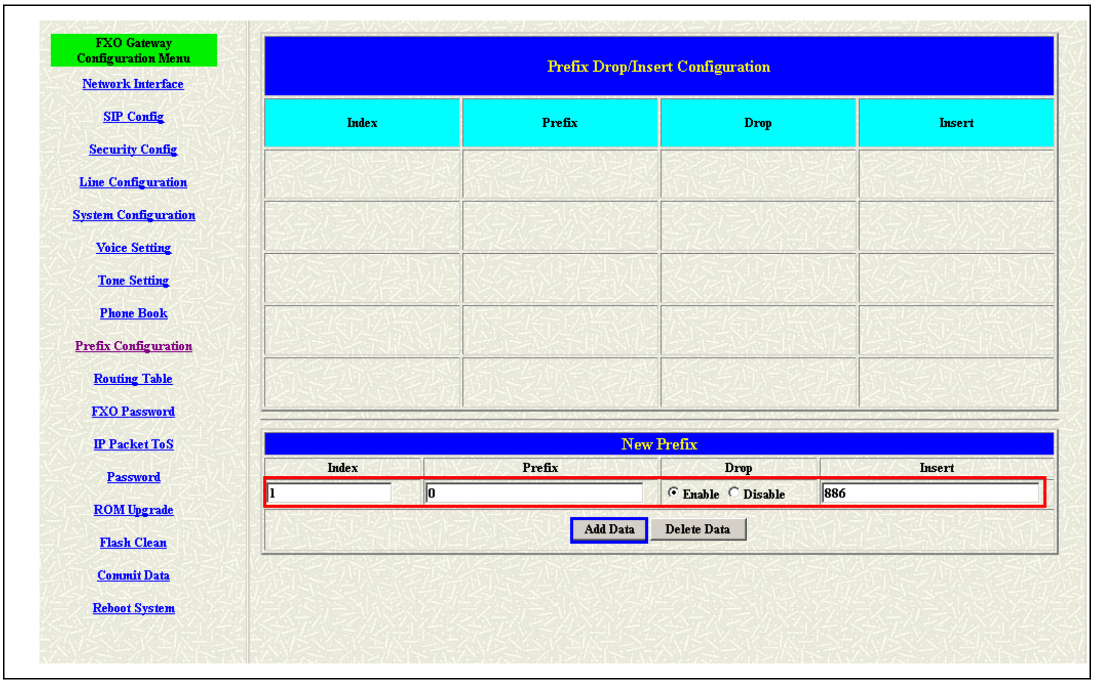


Figure 3.12: Configure the Prefix Table

The usage is as same as the drop, insert function of the Phone Book.

Input (Prefix)	Drop	Insert	Output
100	Disable	X	100
200	Disable	0	0200

300	Enable	X	X
400	Enable	500	500

3 Press the Prefix Configuration button to reload the configuration table (see figure 3.13)

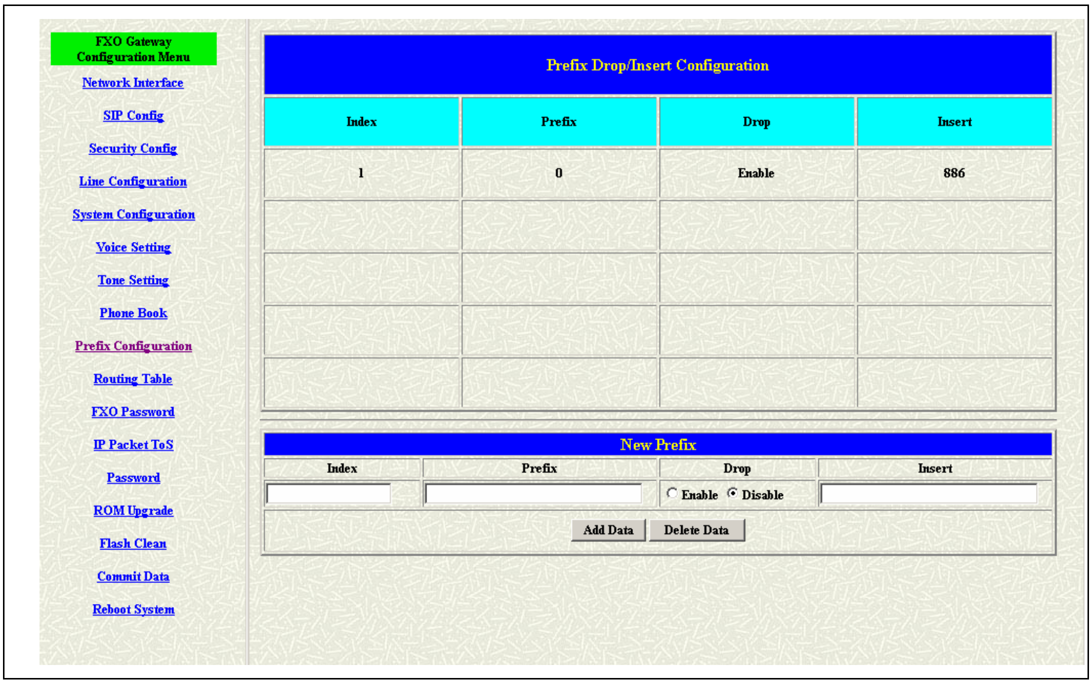


Figure 3.13: Show the added table

4 Please Commit it and Reboot the system if the configuration is finished.

3.10 Routing Table

Routing Table is a rule to define the destination of the calls you make. You could define the rules by the number you dial or by the ports. The Routing Table button will show you the configuration table (see figure 3.14).
In fact, there are two directions of the incoming calls (from IP or FXO side). The explanation of the default routing is as below:

The location with the incoming calls	The location with the destination	The explanation
IP (Default)	Fxo	The destination will be the FXO port when the calls from the IP side without any define rules.
Fxo (Default)	IP	The destination will be the IP side when the calls from the FXO port without any define rules.

The most important usage is for the one-stage-dialing function. For the one-stage-dialing function under the Proxy mode, users have to make sure about that the Proxy could support some kind of the function just like the routing.

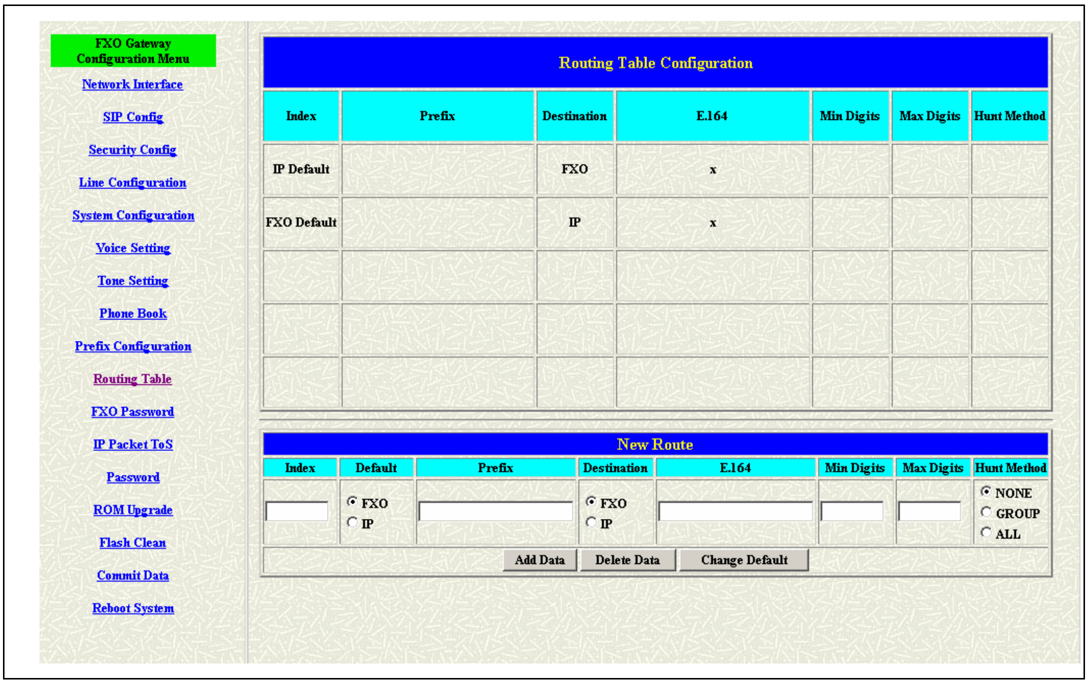


Figure 3.14: Routing Table Configuration

3.10.1 Add a new Routing Table

1 The default setting is changed after you press the Change Default button.

Please press the Routing Table button again to show the new setting. (see figure 3.15)

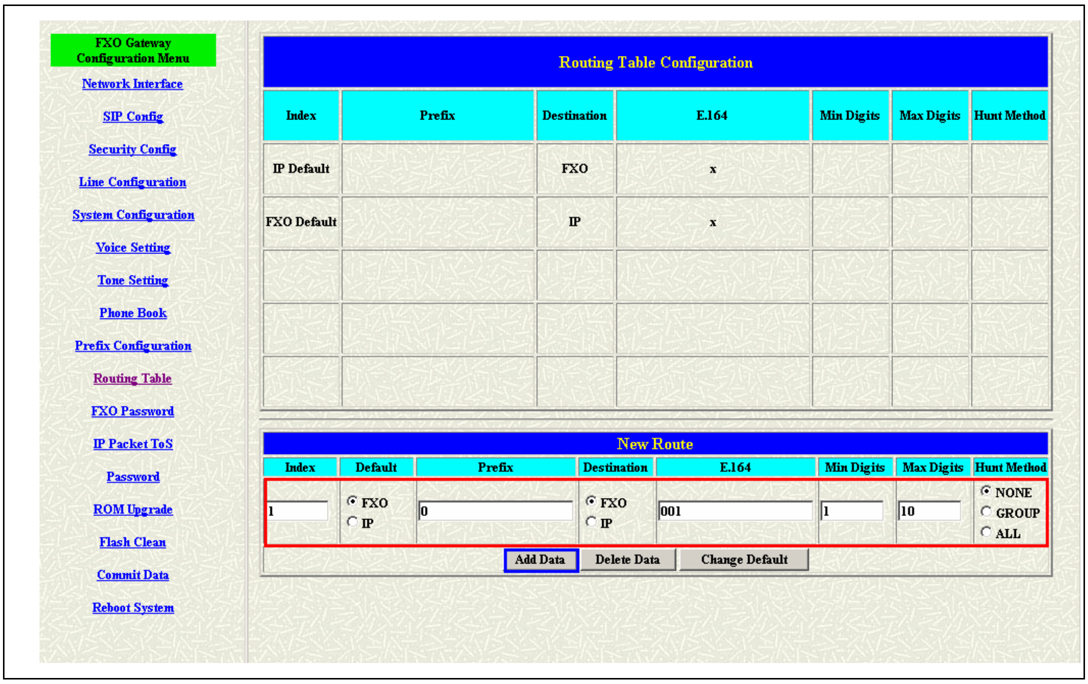


FIGURE 3.15: EDIT AND ADD A NEW ROUTING TABLE

- Index – Define the number of this data.
- Prefix – Define the number you dial. You could just define the first digit of the numbers
- Destination – Define the destination of this rule. There are three directions of the destination.
- E164 – Define a right E164 number of the destination you want.

For example: There are two FXO ports of the gateway (2S2O) and I want the first FXO port (1002 is the default E164 number) to be the destination. So the E164 number I have to define is 1002.

- Min Digits – The minima digits you dial.
- Max Digits – The maxima digits you dial.

The min and max digits are the range for the number you dial. For example: The min digits is 1 and max digits is 10. The call will follow this routing if the number I dial is between 1 and 10 digits. If I dial over 10 digits, this call will

None – Disable this function
Group – The call will search other ports to be the destination with the same group if the origin destination is busy.
All – The call will search other ports to be the destination with the same type if the origin destination is busy.

2 Press Add Data button to save the configuration and press the Routing Table button again to reload the configuration. (see figure 3.16)

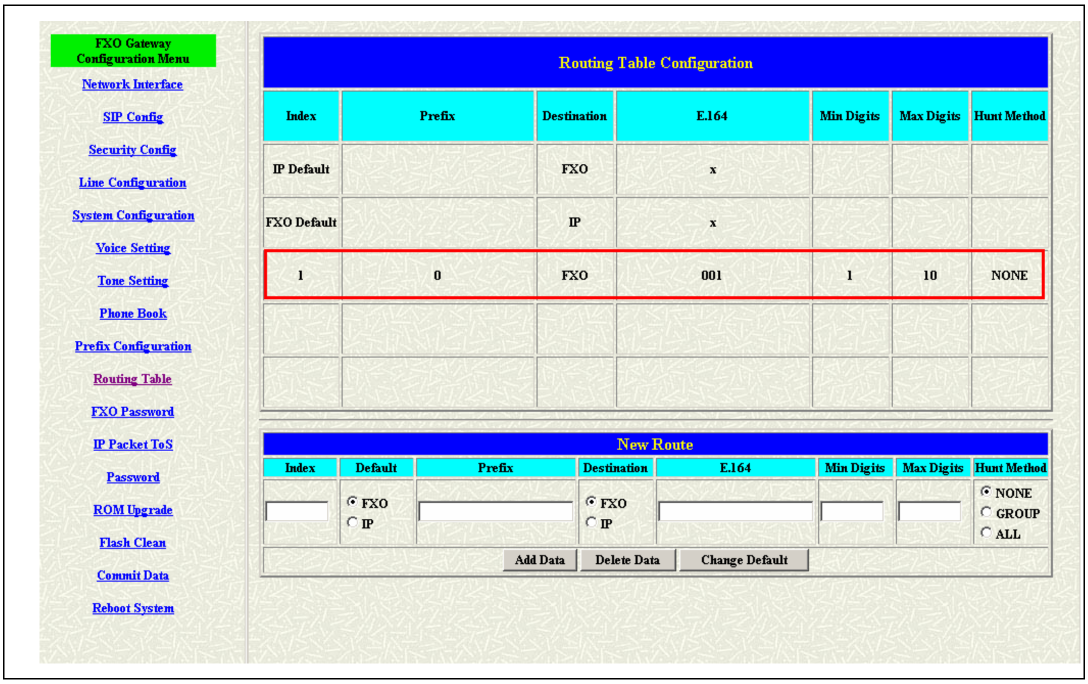


Figure 3.16: New Special Routing

The explanation of figure 3.16 is as below:

When the user dial 0 with the first digit of the numbers (from IP side). The numbers you dial is between 1 and 10 digits. If this call matches the rule, it will be transferred to the FXO port whose E164 number is 001.

3 Please Commit it and Reboot the system if the configuration is finished.

3.11 FXO Password

You will get the IVR if you make calls from PSTN side. The IVR will ask you the password you set, and you could make other calls to IP side if the password you type is correct. Please press the FXO Password button to configure the password (see figure 3.17)

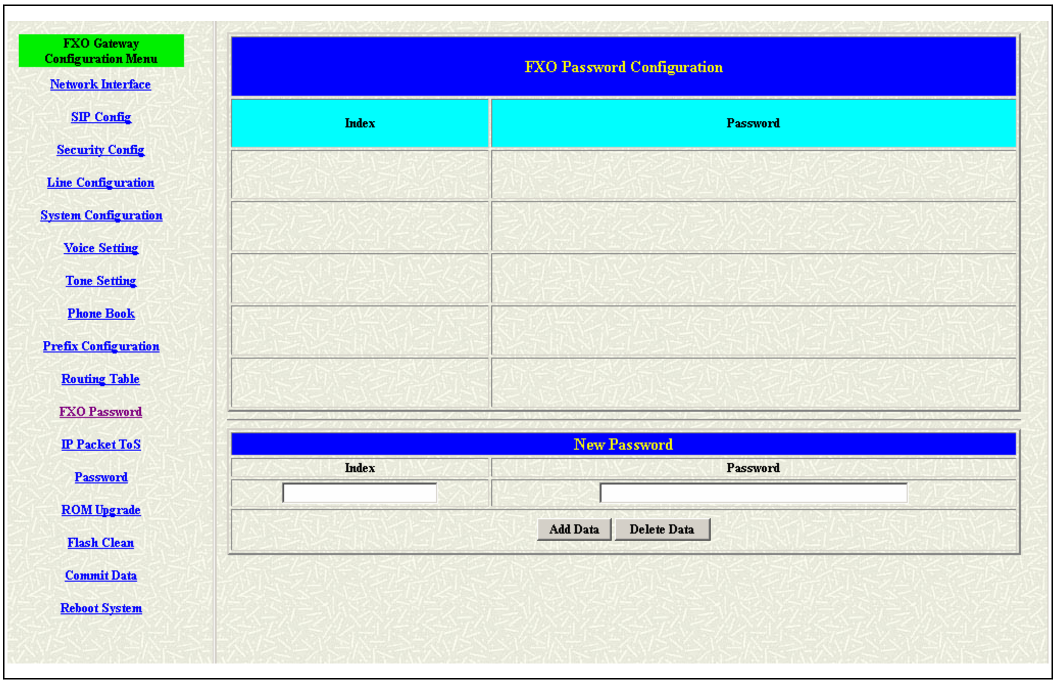


Figure 3.17: FXO Password

- Index – The number of this table.
- Password – The password you set.

This function is only for the calls from the PSTN side. It's not ready for the IP side as so far.

3.12 IP Packet ToS

The Type of Service should be worked with the network router. The router will check all the packets if it support the TOS function. There is a field in the packet for the TOS value. This WEB is for users to configure these values to make the packets with the correct values for the TOS service from the gateway. (see figure 3.18)

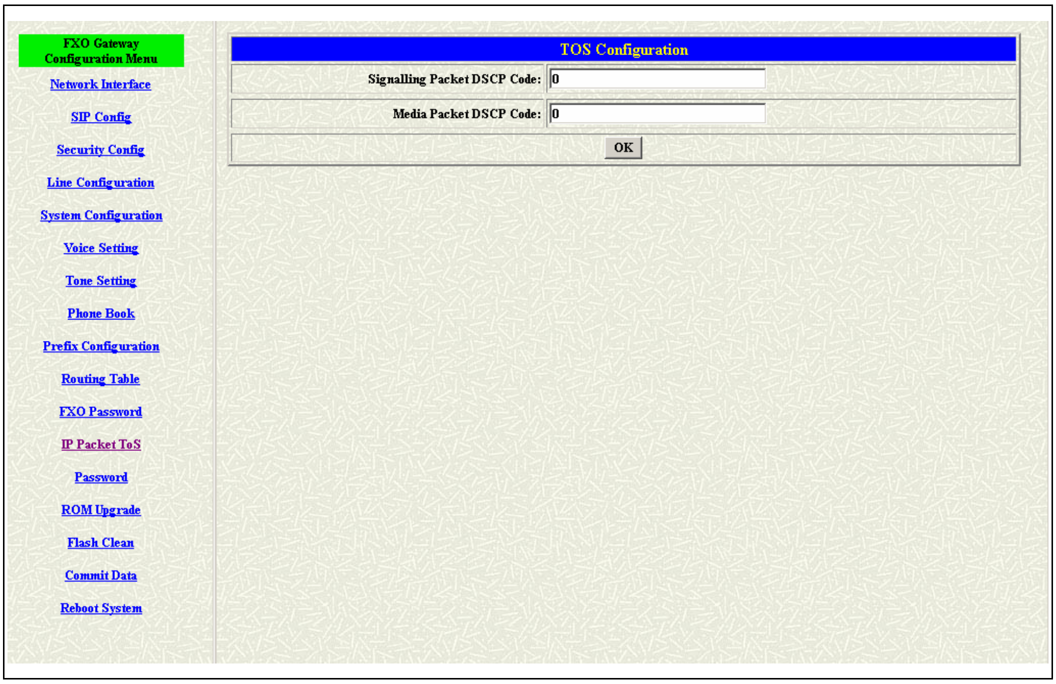


Figure 3.18:TOS Configuration

According to the RFC 1349 document, the TOS value as following :

1000 – minimize delay
0100 – maximize throughput
0010 – maximize reliability
0001 – minimize monetary cost
0000 – normal service

These values are the Binary format. Please change to the Decimal and put these values in to the correct table.

3.13 Password

There are two accounts for login to access or change the configurations. One is “root”, another is “administrator”. Users could define the password for these two login account. The account “root” could make all the configurations back to the default setting, but the account “administrator” couldn’t. This is the difference between these two accounts.

Users could define the password for the accounts in this page. (see figure 3.19)

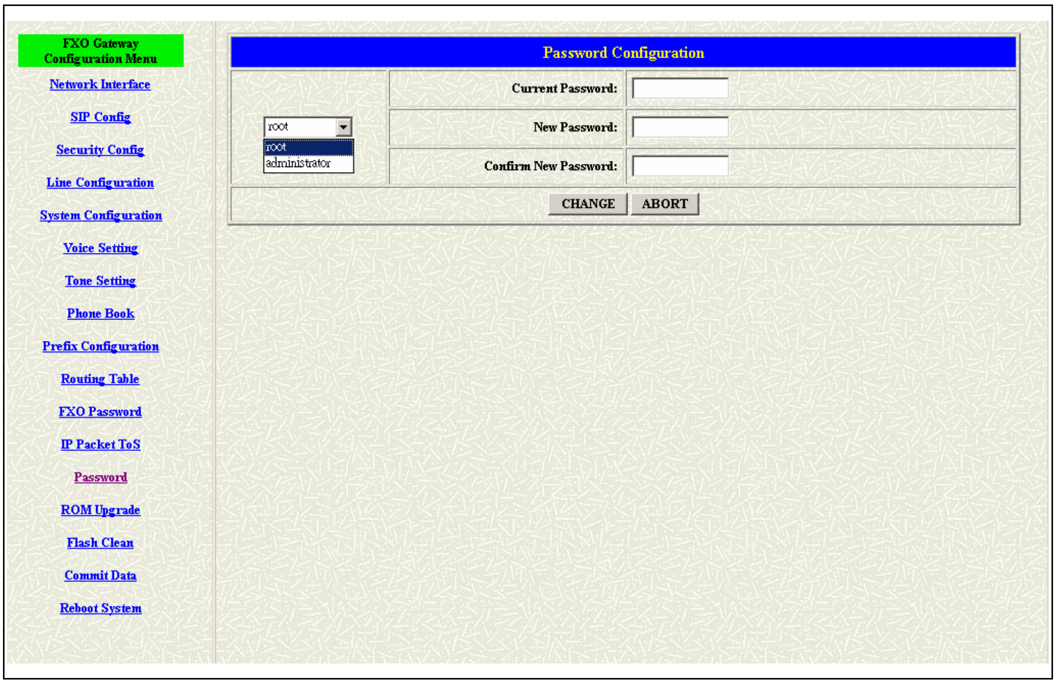


Figure 3.19: Password

- Account – The "root" could make all the configurations back to the default setting except the ip address and the password of the account. But the "administrator" couldn't.
- Current Password – Enter the original password.
- New Password – Enter the new password, which you want.
- Confirm New Password – Enter the new password again.

Please remember the password you configure for the account. It will be more difficult to access it if you forgot the password.

3.14 ROM Upgrade

User could update the firmware just by the web configuration interface. There are two type for the upgrading procedure. One is using the TFTP server, another is using the FTP server. Please follow the step to update the gateway firmware version.

3.14.1 Upgrade using the FTP

1 Pick up the "Rom Upgrade" button to enter the upgrading web page and switch to the FTP method. (see figure 3.20)

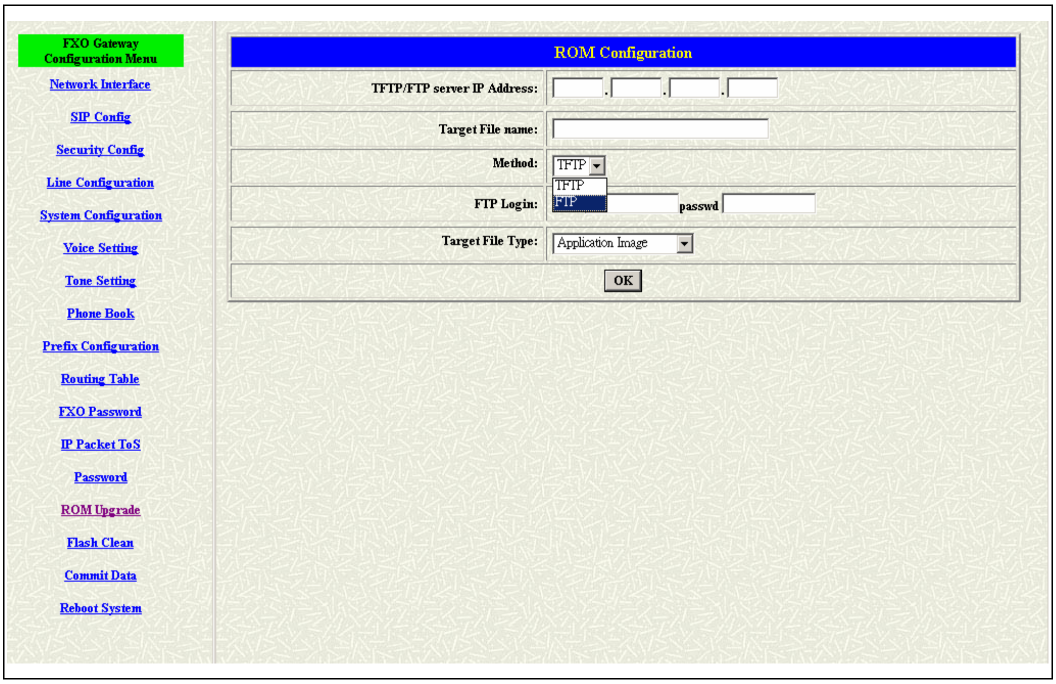


Figure 3.20: ROM Upgrade for FTP

2 Key in the IP address, the login name, password of your FTP server and the correct file name. (see figure 3.21)

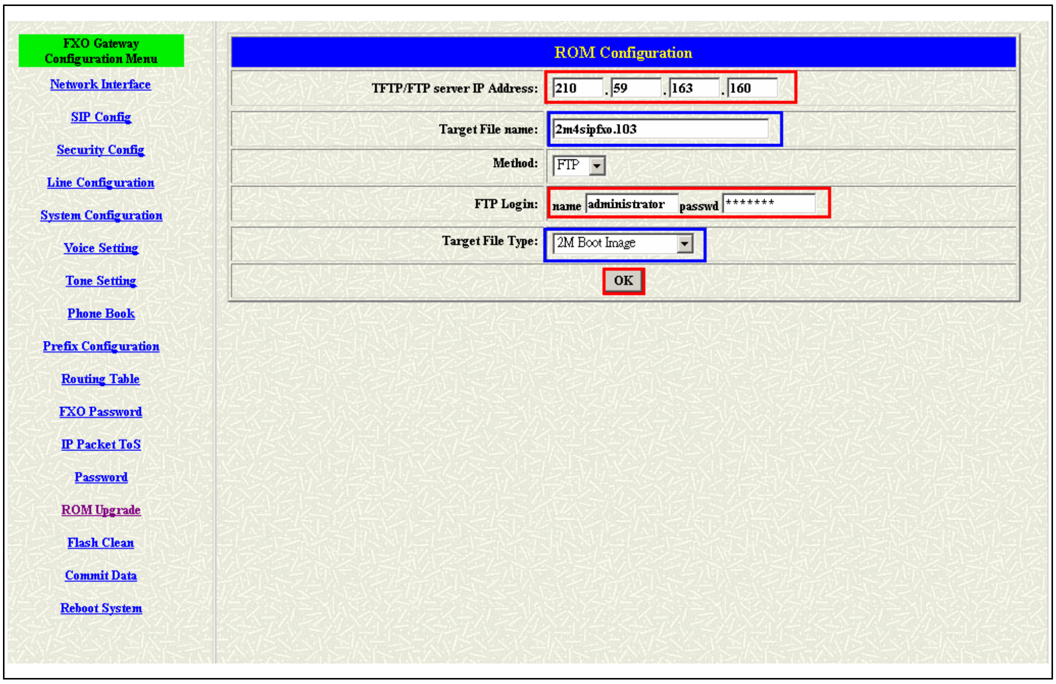


Figure 3.21: FTP information

Please pay more attentions about the blue blank. The Target File Type has to be matched with the Target File name. Please put the correct info about the Target file in this table.

3 Press the OK button to execute the upgrade procedure.

4 Please press the “Reboot System” button to make it reboot.

3.14.2 Upgrade using the TFTP

1 Downloading the TFTP program from our web site and install it first.

Executing the TFTP program before you want to use the TFTP upgrade method.

2 Pick up the "Rom Upgrade" button to enter the upgrading web page and switch to the TFTP method. (see figure 3.22)

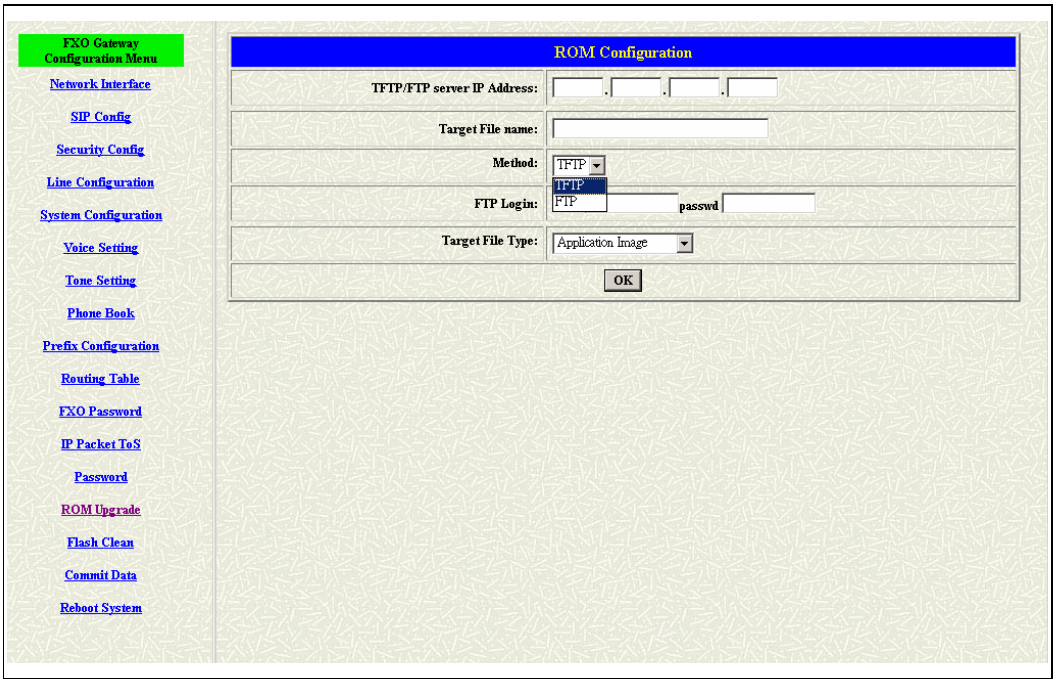


Figure 3.22: ROM Upgrade for TFTP

3 Key in the IP address of your TFTP server, pick up the file type for your upgrade file and the correct file name for upgrading. (see figure 3.23)

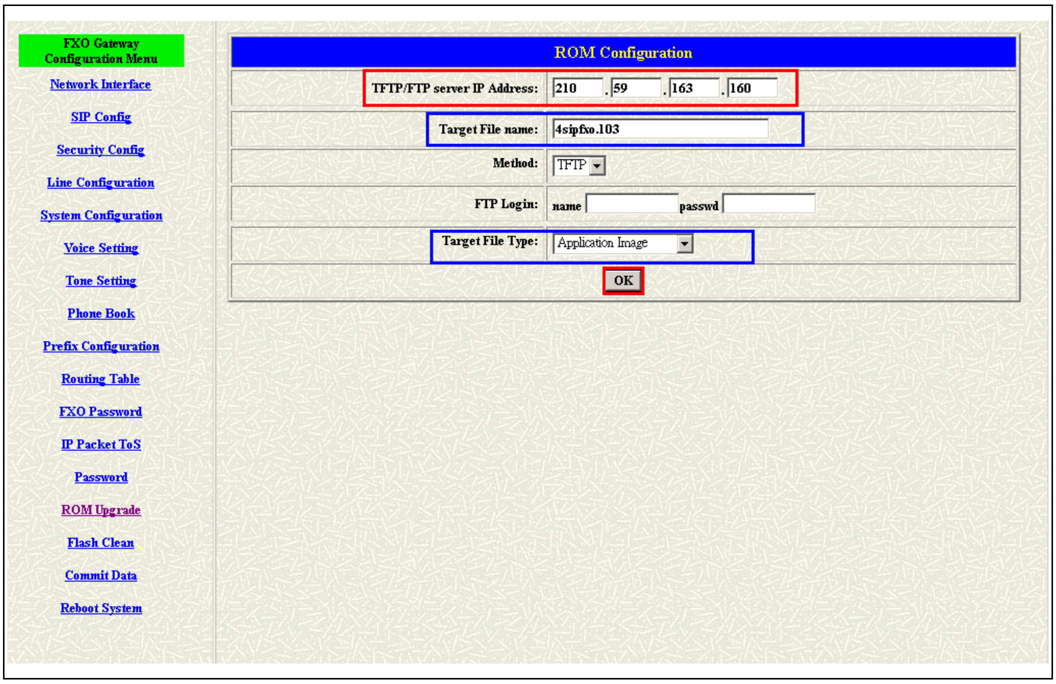


Figure 3.23 : TFTP information

4 Press the OK button to execute the upgrade procedure.

5 Please press the “Flash Clean” button when the procedure is finished.

6 After pressing the “Flash Clean” button, please press the “Reboot System” button to make it reboot.

3.15 Flash Clean

Users could make all the configurations back to the default setting by this button. The password of the account and the networking configuration couldn't be back to the default setting by this command. (see figure 3.24)

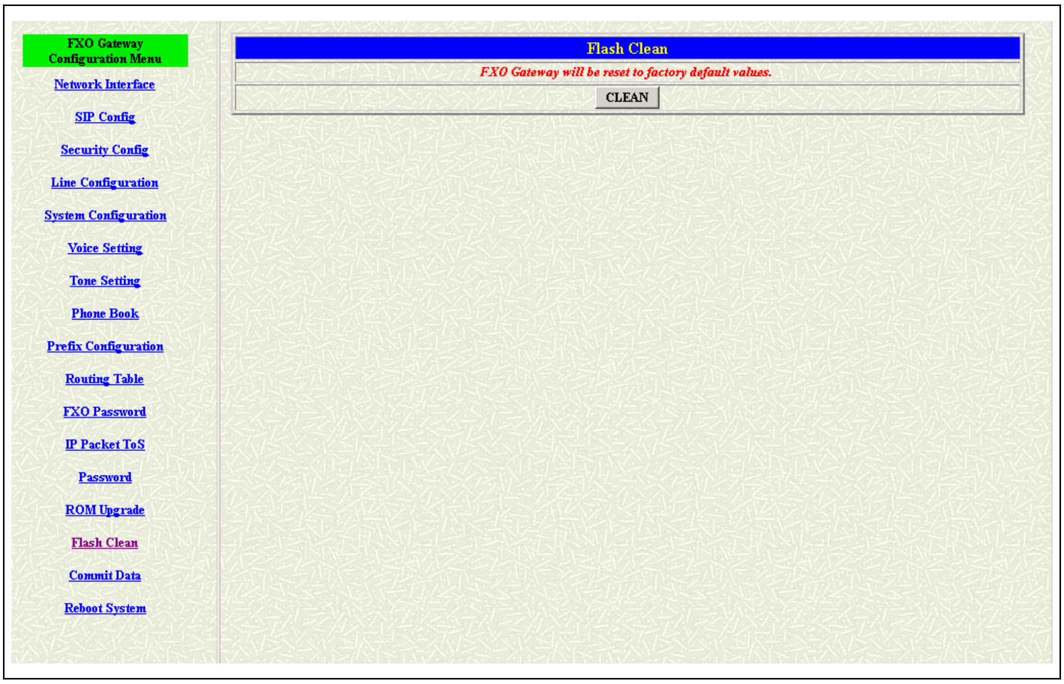


Figure 3.24: Flash Clean

3.16 Commit

This web page could save the configurations if users change some configurations. This is necessary for users change the configurations. (see figure 3.25)

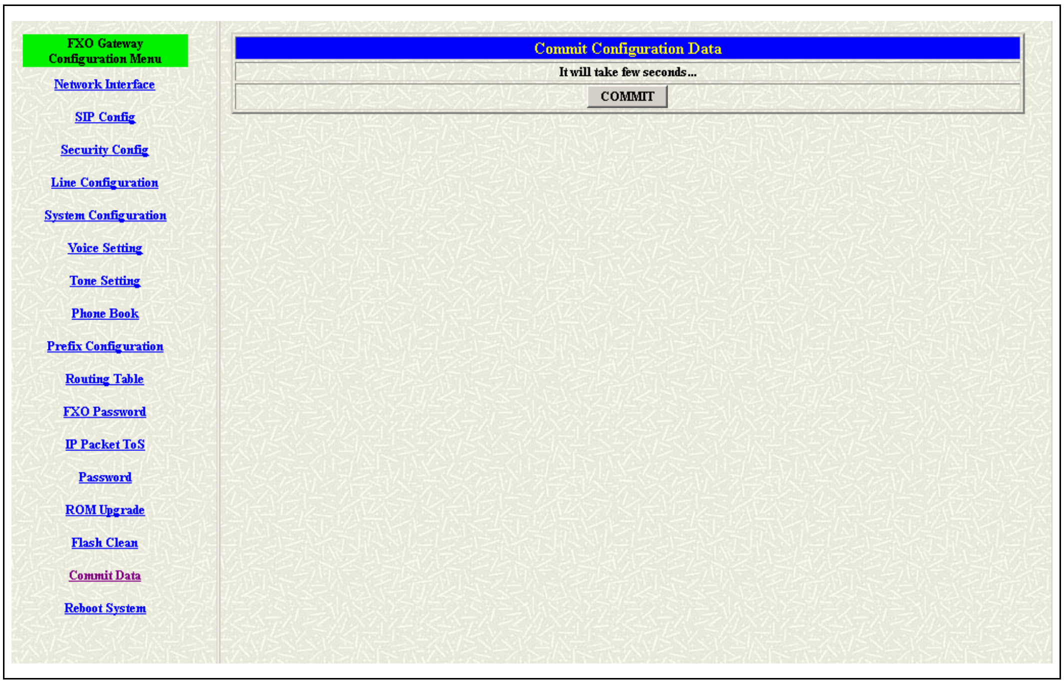


Figure 3.25: Flash Clean

3.17 Reboot

This web page will restart the whole system. This is the necessary step for the changing the configurations and makes it executed. (see figure 3.26)

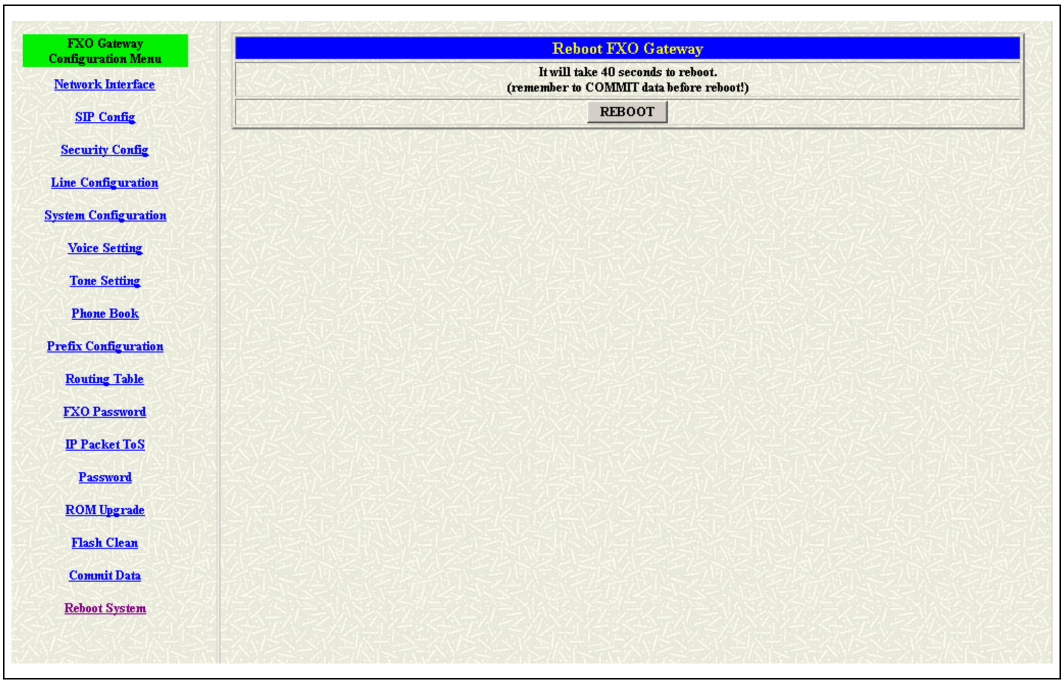


Figure 3.26: Flash Clean

4. Command List

4.1 Hyper Terminal Setting

A terminal emulator is needed when using RS-232 port to configure Gateway. There are kinds of terminal emulator software. Here, we use Microsoft HyperTerminal to depict how to set up terminal emulator:

1. Execute the *Hyper Terminal* program, and then the following windows will pop-up on the screen. (START – Program files – Accessories – Communication – Hyper Terminal)

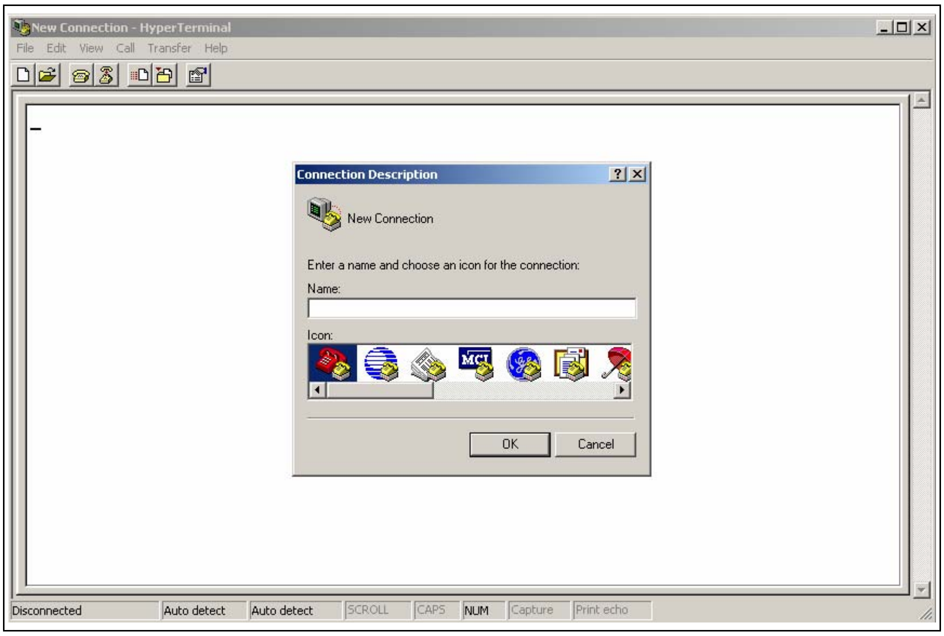


Figure 4.1: Hyper Terminal

2. Define a name for this new connection.

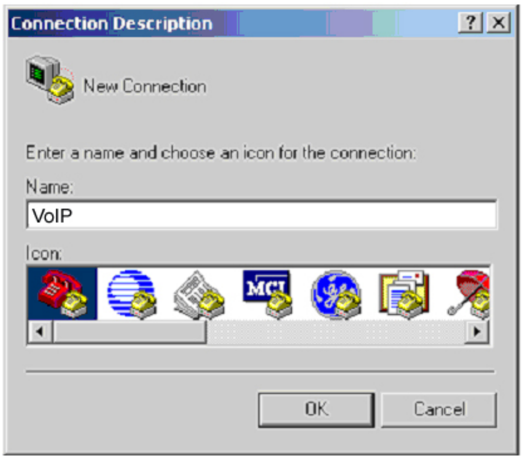


Figure 4.2: Edit the name of the connection

3. After pressing OK button, the next window appear, and then choose ***COM1/2 Port***, which you are going to use.

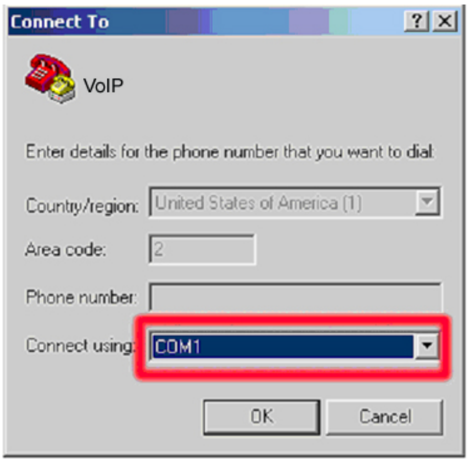


Figure 4.3: Pick up the right interface to use

4. Configure the COM Port Properties as following:
   - Bits per second: 9600
   - Flow control: None

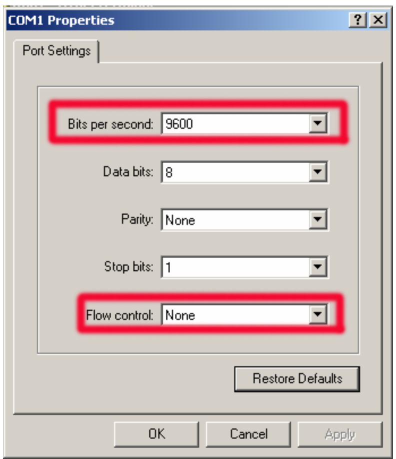


Figure 4.4: Configure the right Bps and control

5. Press 'OK' button, and then start to configure Gateway.

4.2 Command List

4.2.1 [help]

Type **help** or **man** or **?** to list all the available command.

---

usr/config$ help

help	*help/man/? [command]*
quit	*quit/exit/close*
debug	*show debug message*
reboot	*reboot local machine*
flash	*clean configuration from flash rom*
commit	*commit flash rom data*
ifaddr	*Internet address manipulation*
time	*show current time*
ping	*test that a remote host is reachable*
sysconf	*System information manipulation*
sip	*SIP information manipulation*
security	*Security information manipulation*

line	*Line information manipulation*
route	*Routing information manipulation*
prefix	*Prefix drop/insert information manipulation*
pbook	*Phone book information manipulation*
voice	*Voice information manipulation*
tone	*Setup of disconnect tone*
fxopwd	*Setup of FXO password*
record	*Record voice for greeting and ask pin code*
tos	*IP Packet ToS (Type of Service)values*
pt	*DSP payload type configuration and information*
rom	*ROM file update*
passwd	*Password setting information and configuration*

usage: help [command]

4.2.2 [quit]

Type **quit** will quit the Gateway configuration mode and turn back to login prompt (in console mode) or disconnect (in TELNET mode).

```
usr/config$ quit
Disconnecting...
login:
```

Note: It is recommended that type the "**quit**" command before you leave the console. If so, Gateway will ask password again when next user connects to console port.

4.2.3 [debug]

Open debug message will show up specific information while Gateway is in operation. After executing the debug command, it should execute command **debug -open** as well. One example is demonstrated below.

```
usr/config$ debug -add fsm vp
usr/config$ debug -open
```

In this example, user open debug flags including fsm, vp.

Parameters Usage:

-status Display the enabled debug flags.

-add	Add debug flag. *-- fsm: sip related information* *-- vp : voice related information*
-delete	Remove specified debug flag.
-open	Start to show debug messages.
-close	Stop showing debug messages.

4.2.4 [reboot]

After **commit** command, type **reboot** to reload Gateway in new configuration. The procedure is as below:

```
usr/config$ reboot

.Attached TCP/IP interface to cpm unit 0
Attaching interface lo0...done

Hardware auto detect...
Hardware Type : 2FXO REAL_MAXCALL=2

HTTPD initialized...
VoicePacketizermain comming
WorkMode : PROXY_MODE
incoming InitCallArray....REAL_MAXCALL=2
SIP stack was constructed successfully. Version - 2.2.1.8
Start registering to Proxy server

 AC4804[0] is ok
 successful 1 2
 Initialize OSS libraries...OK!
 VP v1.44 stack open sucessfully.

login:
```

4.2.5 [flash]

This command will clean the configuration stored in the flash ROM and reboot Gateway in factory default setting.

Parameter Usage:

-clean clean all the user defined values, and reboot Gateway in factory default mode.

Note: It is recommended that use "flash –clean" after application firmware id upgraded.

4.2.6 [commit]

Save changes after configuring Gateway.

```
usr/config$ commit

This may take a few seconds, please wait....
Commit to flash memory ok!
usr/config$
```

Note: Users shall use ***commit*** *to save modified value, or they will not be activated after system reboot*.

4.2.7 [ifaddr]

Configure and display Gateway network information.

```
usr/config$ ifaddr

LAN information and configuration
Usage:
ifaddr [-print]\|[-mode used]\|[-sntp mode [server][-cmcenter ipaddress]]
ifaddr [-ip ipaddress] [-mask subnetmask] [-gate defaultgateway]
ifaddr [-id username][-pwd password][-http http port]

    -print      Display LAN information and configuration.
    -ip         Specify ip address.
    -mask       Set Internet subnet mask.
    -gate       Specify default gateway ip address
    -mode       Set ip client service(0=Fix ip, 1=DHCP, 2=PPPoE).
    -sntp       Set SNTP server mode and specify IP address.
    -timezone   Set local timezone.
```

-id connection user name for pppoe.

-pwd connection password for pppoe.

-http Http port.

Note:

SNTP mode (0=no update, 1=specify server IP, 2=broadcast mode).

Example:

ifaddr -ip 210.59.163.202 -mask 255.255.255.0 -gate 210.59.163.254

ifaddr -mode 1

ifaddr -sntp 1 210.59.163.254

usr/config$

Parameters Usage:

-print	print out current [ifaddr] settings and status
-ip	assign IP address for Gateway
-mask	assign internet subnet mask
-gate	assign IP default gateway
-mode	Switch the network type (0 = Static IP; 1 = DHCP mode 2 = PPPoE mode)
-sntp	Simple Network Time Protocol (1 = ON; 0 = OFF) When SNTP function is activated, users have to specify a SNTP server as network time source. An example is demonstrated below:
-timezone	set local time zone according to GMT

----

usr/config$ ifaddr -sntp 1 10.1.1.1

10.1.1.1 stands for SNTP server's IP address.

-id	To configure the pppoe connection account for the pppoe connection.
-pwd	To configure the pppoe connection password for the pppoe connection.
-http	To configure http port for the web configuration.

4.2.8 [time]

When SNTP function of Gateway is enabled and SNTP server can be found as well, type **time** command to show current network time.

----

usr/config$ time
Current time is THU JAN 01 05:29:23 1970

4.2.9 [ping]

Use **ping** to test whether a specific IP is reachable or not.
For example: if 192.168.1.2 is not existing while 192.168.1.254 exists.
Users will have the following results:

usr/config$ ping 192.168.1.2
no answer from 192.168.1.2
usr/config$ ping 192.168.1.254

PING 192.168.1.254: 56 data bytes
64 bytes from 192.168.1.254: icmp_seq=0. time=5. ms
64 bytes from 192.168.1.254: icmp_seq=1. time=0. ms
64 bytes from 192.168.1.254: icmp_seq=2. time=0. ms
64 bytes from 192.168.1.254: icmp_seq=3. time=0. ms
----192.168.1.254 PING Statistics----
4 packets transmitted, 4 packets received, 0% packet loss
round-trip (ms)   min/avg/max = 0/1/5
usr/config$

4.2.10 [sysconf]

This command displays system information and configurations.

----

usr/config$ sysconf

System information and configuration
Usage:

sysconf [-idtime digit][-keypad dtmf]
[-rba digit][-eod digit]
[-ring on_time off_time]

sysconf -print

-print	*Display system overall information and configuration.*
-idtime	*Inter-Digits time.(1~10 sec)*
-keypad	*Select DTMF type: 0=In-band, 1=RFC2833.*
-ring	*The ring time for ring detection.(Uint:ms)*
-rba	*The number of ring times before answer.(1~5)*
-eod	*End of dial.(Enable: 1 / Disable : 0)*

Example: sysconf -ring 500
usr/config$

Parameters Usage:

-print	print out all current settings
-idtime	set the duration(in second) of two pressed digits in dial mode as timed out. If after the duration user hasn't pressed next number, it will dial out all number pressed. (1-10 seconds)
-keypad	DTMF replay type. When value is "0", Gateway will transfer DTMF signal via In-Band type, "1" via RFC2833 type. Users can adjust the value according to various applications.
-ring	ring time for ring detection(in ms). When Gateway has incoming call from PSTN side to FXO port, Gateway will determine it is a ring but not noise only if it is longer than this ring time. ***Note:*** *In Taiwan the ring time of PSTN usually is 1000ms, so if user set ring time longer that 1000ms, FXO port may not be able to pick up the call from PSTN side.*
-rba	When the calls from the PSTN side, FXO port will off hook if the ring time is matched with this number.
-eod	It will transfer the DTMF in "#" if users disable the end of dial

function. Users have to press the key pad in “#” if the end of dial function is enable.

4.2.11 [sip]

This command is for sip configuration related parameters.

```
-----------------------------------------------------------------------------------------
usr/config$ sip
sip stack information and configuration
Usage:
    -print          Display SIP stack information and configuration.
    -mode           Configure as Proxy mode or Peer-to-Peer mode.
    -px             Proxy server address. (Proxy IPv4 address or Proxy
                    dns name)
    -domain         Second domain name in the URL (if domain name is
                    not used, specify as null)
    -prefix         Specify prefix string, use it when the UserID contains
                    alphabets
    -line1          Line 1 is E.164 number of L1.
    -line2          Line 2 is E.164 number of L2.
    -expire         The relative time after which the message expires
                    (0~65535).
    -port           SIP local UDP port number (5060~5070). Default :
                    5060
    -rtp            RTP port number (2326~65532). Default : 16384
  Example:
     sip   -px 210.59.163.171 -line1 70 -line2 71 -line3 72 -line4 73

usr/config$
```

Parameters Usage:

-print	print current h323 related settings
-mode	alternatives for proxy or peer-to-peer mode (1=proxy mode; 0=peer-to-peer mode). If users select proxy mode, a valid proxy is needed when Gateway is in operation.

```
-----------------------------------------------------------------------------------------
usr/config$ sip –mode 0 (peer to peer mode)
```

-px	to assign the ip address of the proxy when Gateway is in proxy mode.
-domain	to assign the domain name of the proxy when it is needed.
-prefix	this will be prefix the alphabets before the sip line number.
-line1	assign the port 1 number.
-line2	assign the port 2 number.

Note:

User can also set "x" in line number to disable the port. If the port is disabled, it can only receive calls but not calling out.

Note:

1. This is for 2FXO unit, for 4FXO and 6FXO model, there will be line1 to line4 or line 1 to line6.
2. No matter in Proxy or P2P mode, user only needs to dial line number to reach local port. For example, in P2P mode, user wants to dial from the Line1 to local Line2, only need to dial number of line2.

-expire	It just like the TTL function in H323, the gateway will make sure the registration is success or not for a period times.
-port	define the local sip port for this gateway.
-rtp	to allocate RTP port range—NOT RECOMMENDED. This may be used when RTP port range conflicts with Firewall policy. (each port of Gateway use 2 RTP ports)

4.2.12 [security]

This is the authentication for the SIP account.

----

usr/config$ line

Secuirty information and configuration

Usage:

security [-name username] [-password password]

security -print

-print	*Display system account information and configuration.*
-line	*Specify which line number you want to set the account.*
-name	*Specify user name.*

-password Specify password.

Example:

security -line 1 -name kkk -password 12345

Parameter Usages:

-print	print out all current settings of security.
-line	the line number, which you want to define the security info
-name	the name is as same as the SIP number.
-password	the password for the authentication if it is the necessary for the proxy.

4.2.13 [line]

This command is for configure each line parameters of Gateway.

----

usr/config$ line

Gateway line information and configuration

Usage:

line -config number [the port number]

line -print Gateway line information.

hunt Hunting group.

hotline Hot line configuration.

Example:

line -config 1 hunt 1 hotline 1003

usr/config$

Parameter Usages:

-print	print out all current settings of line
-config	determine which line to configure
-hunt	set hunting group flag of each line. For example, if user assigns the Line1 as hunt group 1, and the Line2 as hunt group 2, they will be determined as 2 different groups. On the other hand, if user assigns the Line1 as hunt group 1, and the Line2 as hunt group 1 too, when having incoming call to the Line1 port, which is busy, this call will be routed to Line2.

-hotline set hotline table. User just dial into the line port of this unit, and gateway will automatically dial out a phone number. In the other hand, user will hear ring back tone or dial tone immediately depended on configurations of destination device. **Note: This function can both work in Proxy or P2P mode.**

Proxy Mode Usage:

Set gateway under proxy mode.

Create a Hotline table with "***line*** " command.

----

usr/config$ line –config 1 hotline 1001

In this example means: if there is a incoming call from PSTN into the port 1, gateway will automatically dial out the number "1001".

P2P Mode Usage:

Set gateway under P2P mode.

Create phone book table with "***pbook*** " command.

Create a Hotline table with "***line*** " command.

----

usr/config$ pbook –add name sipfxo ip 10.1.1.1 e164 1001

usr/config$ line –config 1 hotline 1001

In this example means: if there is a incoming call from PSTN into the port 1, gateway will automatically dial out the number "1001".

4.2.14 [route]

This command is to set routing table for Gateway.

```
usr/config$ route
Routing table information and configuration
Usage:
route -add [prefix number][dst number][e164 number]
          [min number][max number][hunt number]
route -delete index
```

route -modify index [prefix number][dst number][e164 number]
[min number][max number][hunt number]
route -ip [dst number][SIP number]
route -fxo [dst number][SIP number]
route -print Routing table information.
prefix The prefix of dialed number.
dst Destination port(FXO:1/IP:2).
e164 Destination e164 number(when destination is FXO).
min Min digits.(0 ~ 255)
max Max digits.(0 ~ 255)
hunt Hunt method for busy forward(NONE:0 / GROUP:1 / ALL:2)
Example:
route -add prefix 100 dst 1 e164 1001 min 1 max 3 hunt 1
route -ip dst 1 e164 1001
route -fxo dst 1 e164 x
route -modify 1 prefix 100 dst 0 e164 1001 min 1 max 3 hunt 1
route -delete 1
usr/config$

Parameter Usages:

-print	print out all routing table information
-add	add a routing rule in routing table. User can add less than 50 rules. (***route –add prefix "prefix number" dst "destination port type" e164 "SIP number of port" min "minimum digits needed" max "maximum digits can't be exceeded"***)
-delete	delete a routing rule in routing table (***route –delete "index of routing rule"***)
-modify	modify a routing rule in routing table. (***route –modify "index of routing rule" prefix "prefix number" dst "destination port type" e164 "SIP number of port" min "minimum digits needed" max "maximum digits can't be exceeded"***)
-ip	create routing table for incoming call from IP side. (***route –ip dst "destination port type" e164 "SIP number of port"***)
-fxo	create routing table for incoming call from FXO Lines.

(***route –fxo dst "destination port type" e164 "SIPnumber of port"***)

prefix	prefix of the dialed number
dst	destination port, 1 means FXO Lines, 2 means IP side, x means no determinate number.
e164	destination SIP number. This only need to be set when routed port is FXO Lines to determine which port will this call be routed to.
min	minimum digits needed.
max	maximum digits needed.
hunt	set hunt method for busy forward. 0 means no hunting, 1 means hunting method follows the rule of ***[line]***, 2 means hunting method is to hunt between all ports in the same type, for example, destination port is FXO Lines will hunt in all FXO Lines.

Usage Example:

1. ***route –add prefix 100 dst 1 e164 1001 min 1 max 3 hunt 1***
   This command means if gateway has incoming call's prefix number is 100, and total digits are between 1 to 3, this call will be routed to FXO port whose number is 1001. If the destination port is busy, call will be routed to another port, which is in the same group.
2. ***route –ip dst 1 e164 1002***
   This command means incoming call from IP side will be routed to FXO Line of number 1002.

4.2.15 [prefix]

This command is for make rules for drop or insert prefix digits.

----

usr/config$ prefix

Prefix drop/insert information and configuration
Usage:
prefix -add [prefix number][drop number][insert digits]
prefix -delete index
prefix -modify index [prefix number][drop number][insert number]
prefix -print   Prefix drop/insert information.

prefix The prefix of dialed number.
drop Drop prefix(Enable:1/Disable:0).
insert Insert digits.

Example:

prefix -add prefix 100 drop 1 insert 2000
prefix -add prefix 100 drop 1
prefix -add prefix 100 drop 0 insert 200
prefix -delete 1
prefix -modify 1 prefix 100 drop 0 insert 300

usr/config$

Parameter Usages:

-add add a rule to drop or insert prefix digits of incoming call.(***prefix –add prefix "prefix number" drop 0/1 insert "insert number"***)

-delete delete a rule to drop or insert prefix digits of incoming call. (***prefix –delete prefix "prefix number"***)

-modify modify a rule to drop or insert prefix digits of incoming call. (***prefix –modify prefix "prefix number" drop 0/1 insert "insert number"***)

prefix set which prefix number to implement prefix rule.

drop enable or disable drop function.If this function is enabled, Gateway will drop prefix number on incoming call.

insert set which digit to insert on incoming call.

4.2.16 [pbook]

Phone Book function allows users to define their own numbers, which mapping to real IP address. It is effective only in peer-to-peer mode. When adding a record to Phone Book, users also have to reboot the machine after the commit command, and the record will be effective immediately.

----

usr/config$ pbook

Phone book information and configuration

Usage:

pbook [-add [name string][e164 number][ip address]

[port number][drop digit][insert number]]
[-modify number [name string][e164 number][ip address]
[port number][drop digit][insert number]]
[-delete number]
pbook -print

-print *Display phone book information and configuration.*
-add *Add new phone book record)*
-delete *Delete phone book record*
-modify *Modify phone book record.*
name : 1 ~ 10 characters.
e164 : 1 ~ 10 digits.
ip : IP adress.
port : 1024 ~ 65535.
drop : 0:Disable/1:Enable.
insert : 1 ~ 10 digits.

Example:
pbook -add name test e164 1234 ip 192.168.1.10 drop 1 insert 5678
pbook -delete 1
pbook -modify 1 name test e164 5678 ip 192.168.1.10 drop 0

usr/config$

Parameter Usages:

-print print out current contents of Phone Book. (***pbook -print***) Users can also add *index number*, from 1 to 100, to the parameter to show specific phone number. (Ex. ***pbook –print 1***)

Note: <index number> means the sequence number in phone book. If users do request a specific index number in phone book, Gateway will give each record a automatic sequence number as index.

-add add a new record to phone book. When adding a record, users have to specify ***name***, ***ip***, and ***e164*** number to complete the command.

name	name to represent callee.
e164	The SIP number for mapping with IP address of called
ip	ip address of called
drop	drop e.164 number when dial out. 0 means to keep e.164 number, 1 means to drop e.164 number when dialing out.
insert	insert digits.(1~10 digits)
-delete	delete a specific record. “pbook –delete 3” means delete **index 3** record.
-modify	modify an existing record. When using this command, users have to specify the record’s index number, and then make the change.

PhoneBook Rules:

The SIP number defined in phone book will fully carry to destination. It is not just a representative number for destination’s IP Address. In other words, user dial this number to reach the destination, destination will receive the number and find out if it is matched to itself, including Line number in some particular device.

4.2.17 [voice]

The voice command is associated with the audio setting information.
There are four voice codecs supported by Gateway.

----

usr/config$ voice

Voice codec setting information and configuration
Usage:
voice [-send [G723 ms] [G711A ms] [G711U ms] [G729 ms] [G729A ms] [G729B ms] [G729AB ms]]
[-volume [voice level] [input level] [dtmf level]]
[-nscng [G711U used1] [G711A used2] [G723 used3]]
[-echo used] [-mindelay t1] [-maxdelay t2] [-optfactor f]
voice -print
voice -priority [G723] [G711A] [G711U] [G729] [G729A] [G729B] [G729AB]

-print Display voice codec information and configuration.

-send Specify sending packet size.
G.723 (30/60 ms)
G.711A (20/40/60 ms)
G.711U (20/40/60 ms)
G.729 (20/40/60 ms)
G.729A (20/40/60 ms)
G.729B (20/40/60 ms)
G.729AB (20/40/60 ms)
-priority Priority preference of installed codecs.
G.723
G.711A
G.711U
G.729
G.729A
G.729B
G.729AB
-volume Specify the following levels:
voice volume (0~63, default: 29,28),
input gain (0~63, default: 26),
dtmf volume (0~31, default: 23),
-nscng No sound compression and CNG. (G.723.1 only, On=1, Off=0).
-echo Setting of echo canceller. (On=1, Off=0, per port basis).
-mindelay Setting of jitter buffer min delay. (0~150, default: 90).
-maxdelay Setting of jitter buffer max delay. (0~150, default: 150).

Example:
voice -send g723 60 g711a 60 g711u 60 g729 60 g729a 60 g729b 60 g729ab 60
voice -volume voice 20 input 32 dtmf 27
voice -echo 1 1
usr/config$

Parameters Usage:

-print print current voice information and configurations.

-send define packet size for each codec. 20/40/60ms means to send a voice packet per 20/40/60 milliseconds. The smaller the packet size, the shorter the delay time. If network is in

good condition, smaller sending packet size is recommended. In this parameter, 20/40/60ms is applicable to G.711u/a law, and G.729/G.729A/G.729B/G.729AB codec, while 30/60ms is applicable to G.723.1 codec.

-priority codec priority while negotiating with other h323 device. This parameter determines the listed sequence in h.245 TCS message. The codec listed in left side has the highest priority when both parties determining final codec. User can also select the particular codec without others.

----

usr/config$ voice –priority g723 (only select this codec)
usr/config$ voice –priority g723 g729 g711u g711a (select four codecs, and g723 is the first choice)

-volume There are three adjustable value. **voice volume** stands for volume, which can be heard from Gateway side; **input gain** stands for volume, which the opposite party hears; **dtmf** volume stands for DTMF volume/level, which sends to its own Line.

Note: level of volume is too high or too low may be result in bad performance while connecting to each other.

-nscng silence suppression and comfort noise generation setting (1 = ON; 0 = OFF). It is applicable to G.723 codec only. An example is demonstrated below:

----

usr/config$ voice -nscng g723 1

-echo activate each canceler (1 = ON; 0 = OFF).
-mindelay the minimum jitter buffer size. (Default value= 90 ms)
-maxdelay the minimum jitter buffer size. (Default value= 150 ms)

----

usr/config$ voice –mindelay 90 –maxdelay 150

---

*Note: be sure to know well the application before you change **voice** parameters because this might cause incompatibility.*

4.2.18 [tone]

This command is basically for FXO ports.

----

usr/config$ tone

Disconnect tone and remote ring back tone configuration

Usage:

tone [num][freqHi][freqLo][freqHiLev][freqLoLev]
[Tone1ON][Tone1OFF][Tone2ON][Tone2OFF]]

tone -print Display disconnect tone configuration.

[num] Tone index(1~4:Disconnect tone / 5~8:Remote ring back tone).

Example:

tone -print

tone 1 620 480 8 8 50 50 1023 1023

usr/config$

Parameter Usages:

-print show all tone configuration

[num] tone index. 1~4 is disconnect tone, 5~8 is remote ring back tone.
For FXO ports Gateway must detect disconnect tone to determine when to disconnect the call, so user must set disconnect tone of PBX or PSTN network connected to FXO ports.
When making a call from FXO ports, there are 2 ways to detect callee has already picked up the call, one is to detect reverse signal, the other is to detect the termination of ring back tone, so user must set ring back tone of PBX or PSTN network.
(If user doesn't know about the frequency of disconnect tone

or ring back tone, please refer to ***[record]*** command to detect frequency.)

For each tone may has 1 set or 2 sets (high and low) of frequencies. If user wants to set 0 in on/off time, please set "1023" represent "0". (ex. ***tone 1 620 480 8 8 50 50 1023 1023***)

(***tone "index of tone" "frequency of high" "frequency of low" "level of high" "level of low" "on time of high" "off time of high" "on time of low" "off time of low"***)

4.2.19 [fxopwd]

This command is for FXO ports.

----

usr/config$ fxopwd

FXO password information and configuration

Usage:

fxopwd -add [passwd number][direction number]

fxopwd -delete   index

fxopwd -modify   index [passwd number][direction number]

fxopwd -print   FXO password information.

passwd   The password.

Example:

fxopwd -add passwd 1234

fxopwd -delete 1

fxopwd -modify 1 passwd 1234

usr/config$

Parameter Usages:

-print	show all FXO password configuration
-add	add 1 set of FXO password
-delete	delete 1 specific set of FXO password
-modify	modify 1 specific set of FXO password
passwd	password

4.2.20 [record]

User can record greeting and askpin file and analyze tone frequency by calling in FXO line of Gateway.

```
-------------------------------------------------------------------------------------------
usr/config$ record

Recoed greeting voice and ask pin code voice, tone analize.
Usage:
record -greeting filename
          -askpin filename
          -tone
Example:
       record -greeting greeting.100
       record -askpin askpin.100
       record -tone

usr/config$
```

Parameter Usages:

-greeting   record greeting file. User must assign a file name for greeting, once record is finished, file recorded will be display in rom –print.

```
-------------------------------------------------------------------------------------------
usr/config$ record -greeting test.100


Please off hook TEL 1 and press (N) for next step...
( Please make calls from the PSTN side into this port )
n

Press (R) to start record...
r

Press (S) to stop record...
........................................................................................
```

...
...s........
...
....

Press (P) to play the voice or (W) to write to flash or (Q) to quit...
p
w

Please wait a moment...

Write flash ok...

Boot Rom : sdboot.200
Application Rom : 4sipfxo.103
DSP App : 48302ce3.300
DSP Kernel : 48302ck.300
DSP Test Code : 483cbit.bin
Greetings : test.100
Ask Pin : askpin.100
q
usr/config$

-askpin record askpin file. User must assign a file name for askpin file, once record is finished, file recorded will be display in rom –print.

----

usr/config$ record -askpin askpintest

Please off hook TEL 1 and press (N) for next step...

(Please make calls from the PSTN side into this port)

n

Press (R) to start record...

---

r

Press (S) to stop record...
...
...
...
...
...
.........................s...
...
...
.
Press (P) to play the voice or (W) to write to flash or (Q) to quit...
p
w

Please wait a moment...

Write flash ok...

Boot Rom : sdboot.200
Application Rom : 4sipfxo.100
DSP App : 48302ce3.300
DSP Kernel : 48302ck.300
DSP Test Code : 483cbit.bin
Greetings : greeting.100
Ask Pin : askpintest
q
usr/config$

Note: Remember to press enter after press any command.

-tone analyze tone frequency. Gateway can analyze tone frequency as user provide tone in FXO Line1.

----

usr/config$ record -tone

Press (R) to start record...
r
...
...
...
....

Analyzing!! Please wait a moment……..
Frequency 1 : 480
Frequency 2 : 620
Frequency 3 (2623) is more than 1000, please ignore it.
0.25sec on 0.25sec off
usr/config$

Note:

1. ***About the tone detection or tone recording for FXO unit, two extension or PSTN line is necessary.***
2. ***Records disconnect tone: Please read the procedure of recording disconnect tone file from the web site in application.***
3. ***The values of disconnect tone and ring back tone will not write in flash automatically. Please use the command in "tone" to write in the tone table.***

The Procedures of recording the disconnect tone

Before you start ：

Prepare two PSTN lines, which connect with the Line 1 and Line 2 port.

Please record the disconnect tone just follow the stage as below ：

1. Please enter the command before you record the disconnect tone：

   record –tone

2. Make a call from PSTN side into Line 2 port.
3. You will get a greeting when the call enters the gateway.
4. Pease dial the number of the Line 1 port.
5. Users will get the dial tone from the PSTN side and please dial the number to contact with another person.
6. Please drop the call from the calling aide and the called side could get the disconnect tone from the Line 2 port.
7. When you get the disconnect tone from the Line 2 port, press <**R**> and <**ENTER**> buttons to start recording the disconnect tone.
8. Please hang up another side if users get the message as below：

   Analizing!! Please wait a moment…
9. There are three values you will get after analyzing. Please leave the value which is over 1000 Hz, this is not the frequency of disconnect tone.
10. Please put the frequency in the tone table just follow the command：

    tone 4 420 680 8 8 25 25 50 50

【Example-1】

(Make a call from PSTN to FXO port)
usr/config$ record -tone

Press (R) to start record...
(Please make sure that you are already finish the steps 2 ~ 7)
r (Press “Enter” button after you key in “R”)
....
....
....
....

Analizing!! Please wait a moment...
(You coule hang up the call from PSTN if you get this message)

Frequency 1 : 481

Frequency 2 (2623) is more than 1000, please ignore it.
Frequency 3 : 621

tone 4 481 621 8 8 25 25 1023 1023
(Put this value in to the tone table)

tone –print

Disconnect tone 1 paramter

Frequency high	: 620
frequency low	: 480
frequency high level	: 8
frequency low level	: 8
Tone1 on	: 25
Tone1 off	: 25
Tone2 on	: 1023
Tone2 off	: 1023

Disconnect tone 2 paramter

Frequency high	: 450
frequency low	: 0
frequency high level	: 8
frequency low level	: 0
Tone1 on	: 35
Tone1 off	: 35
Tone2 on	: 1023
Tone2 off	: 1023

Disconnect tone 3 paramter

Frequency high	: 620
frequency low	: 480
frequency high level	: 8
frequency low level	: 8
Tone1 on	: 50
Tone1 off	: 50
Tone2 on	: 1023
Tone2 off	: 1023

Disconnect tone 4 paramter

Frequency high	: 621

frequency low : 481
frequency high level : 8
frequency low level : 8
Tone1 on : 25
Tone1 off : 25
Tone2 on : 50
Tone2 off : 50
(Confirm the values is correct or not)

(Key in the commit and reboot command if you finish the procedures as above)

【Example-2】

(Make a call into FXO port)
usr/config$ record -tone

Press (R) to start record...
(Please make sure that you are already finish the steps 2 ~ 7)
r (Press "Enter" button after you key in "R")
....
....
....
....

Analizing!! Please wait a moment...
(You coule hang up the call from PSTN if you get this message)

Frequency 1 : 473
Frequency 2 (2623) is more than 1000, please ignore it.
Frequency 3 (1856) is more than 1000, please ignore it.

tone 4 473 473 8 8 25 25 1023 1023
(Please configure the high and low frequency as the same value if you just get a singal frequency)

tone –print

Disconnect tone 1 paramter

Frequency high	: 620
frequency low	: 480
frequency high level	: 8
frequency low level	: 8
Tone1 on	: 25
Tone1 off	: 25
Tone2 on	: 1023
Tone2 off	: 1023

Disconnect tone 2 paramter

Frequency high	: 450
frequency low	: 0
frequency high level	: 8
frequency low level	: 0
Tone1 on	: 35
Tone1 off	: 35
Tone2 on	: 1023
Tone2 off	: 1023

Disconnect tone 3 paramter

Frequency high	: 620
frequency low	: 480
frequency high level	: 8
frequency low level	: 8
Tone1 on	: 50
Tone1 off	: 50
Tone2 on	: 1023
Tone2 off	: 1023

Disconnect tone 4 paramter

Frequency high	: 621
frequency low	: 481
frequency high level	: 8
frequency low level	: 8
Tone1 on	: 25
Tone1 off	: 25
Tone2 on	: 50
Tone2 off	: 50

(Confirm the values is correct or not)

(Key in the commit and reboot command if you finish the procedures as above)

4.2.21 [tos]

IP Packet ToS(type of Service)/Differentiated Service configuration.

----

usr/configtos

IP Packet ToS(type of Service)/Differentiated Service configuration

Usage:

tos [-rtptype dscp]

tos [-sigtype dscp]

tos -print

[-rtpreliab mode]

tos -print

Example:

tos -rtptype 7 -sigtype 0

Parameter Usages:

-rtptype the packages of voice

-sigtype the package of call signal

Note:

The value of rtptype and sigtype is from 0 to 63.

It's working if it supported by your network.

4.2.22 [pt]

RTP payload type configuration and information

----

usr/config$ pt

RTP payload type configuration and information

Usage:

pt-print Display the RTP payload type information

-rfc2833 Configure the DTMF RFC2833 payload type

-dtmf Configute the DTMF payload type
-fax Configure the FAX payload type
-faxbypass Configure the FAX ByPass payload type
-modembypass Configure the MODEM ByPass payload type
-redundancy Configure the Redundancy payload type
-modemrelay Configure the MODEM Relay payload type

Example:
pt -rfc2833 96 -fax 101

usr/config$

Users could configure the payload type for this function.

4.2.23 [rom]

ROM file information and firmware upgrade function.

----

usr/config$ rom

ROM files updating commands
Usage:
rom [-print][-app][-boot][-dsptest][-dspcore][-dspapp][-greet][-askpin]
-s TFTP/FTP server ip -f filename
rom -print
-print show versions of rom files. (optional)
-app update main application code(optional)
-boot update main boot code(optional)
-boot2m update 2M code(optional)
-dsptest update DSP testing code(optional)
-dspcore update DSP kernel code(optional)
-dspapp update DSP application code(optional)
-greeting update greeting voice file(optional)
-askpin update ask pin code voice file(optional)
-s IP address of TFTP/FTP server (mandatory)
-f file name(mandatory)
-method download via TFTP/FTP (TFTP: mode=0, FTP: mode=1)

-ftp specify username and password for FTP

Note:

This command can run select one option in 'app', 'boot',
, 'dsptest', 'dspcore', and 'dspapp'.

Example:

rom -method 1
rom -ftp vwusr vwusr
rom -app -s 192.168.4.101 -f app.bin

usr/config$

Parameter Usages:

-print show versions of all rom files

-app, boot, boot2m, dsptest, dspcore, dspapp, greeting, askpin to update main Application program code, Boot code, DSP testing code, DSP kernel code, DSP application code, greeting file, askpin file.

-s to specify TFTP server's IP address when upgrading ROM files.

-f to specify the target file name, which will replace the old one.

-method to decide using TFTP or FTP as file transfer server. "0" stands for TFTP, while "1" stands for FTP.

-ftp if users choose FTP in above item, it is necessary to specify pre-defined username and password when upgrading files.

4.2.24 [passwd]

For security concern, users have to input the password before entering configuration mode. "**passwd**" command is for password setting purpose.

----

usr/config$ passwd

Password setting information and configuration

Usage:

passwd -set Loginname Password

---

passwd -clean

Note:

1. Loginname can be only 'root' or 'administrator'

2. passwd -clean will clear all passwd stored in flash,
please use it with care.

Example:

passwd -set root Your_Passwd_Setting

usr/config$

Parameter Usages:

-set

(***passwd –set "login name" "password"***)

Note : "login name" can be "***root***" or "***administrator***" only. "root" and "administrator" have the same authorization, except some commands that can be executed by "root" only – "***passwd –clean***", "***rom –boot***", "***rom –bot2m***" and "***flash –clean***".